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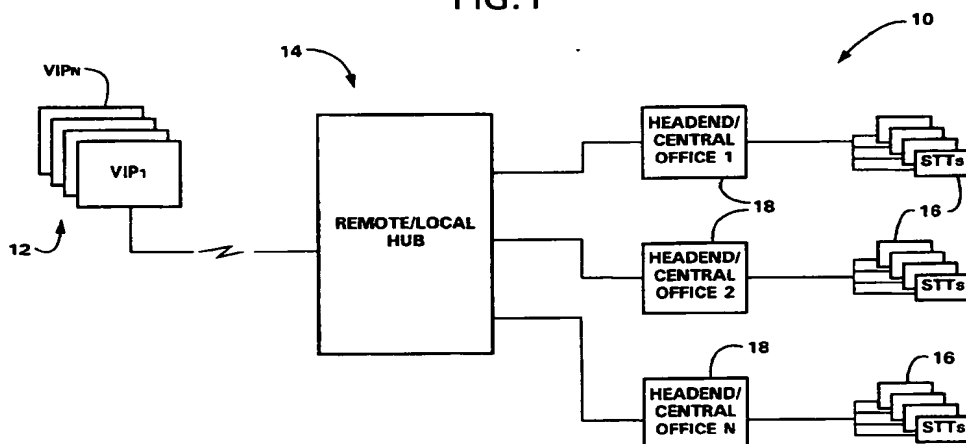
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(54) Distributed system architecture for digital broadcast and interactive services

(57) A cable television communication system network (10) comprising a remote hub (14) which is in communication with a plurality of head ends/central office (18) via a fibre optic interface, a plurality of user terminals (16) associated with each head end via a hybrid fibre coax interface and a plurality of interactive and broadcast video information providers (VIPs) (12) which are in communication with the remote hub (14). The remote hub includes means for formatting broadcast signals received from selected VIPs of the broadcast VIPs (12) for direct transmission to the head end/user

terminal interfaces such that the user terminals (16) receive said formatted broadcast signals without reformatting by said headends, means for selectively controlling distribution of communication signals between said interactive VIPs and said headends (18), means for tracking usage of VIP services by said user terminals for at least billing purposes. Each said headend (18) means for facilitating interactive VIP communications between its associated user terminals (16) and said VIPs (12).

FIG. 1



Description

This invention relates generally to cable television communication systems. More particularly, the invention relates to a cable television communication network architecture that allows distributed delivery of broadcast and interactive digital services over a hybrid fiber-coax metropolitan area network to consumer residences.

The tremendous growth in the use of computer networks and electronic databases over the past decade has resulted in the accumulation of a wealth of electronic information. In order to remain competitive, businesses and individuals increasingly have sought access to this electronic information via on-line services and databases. Although the diversity of electronic interactive services is expected to increase, the means to transmit the information to businesses and homes has remained rather limited. Most of the limitations involve the physical infrastructure that existed in the past and to some extent still exists. Existing upgraded coaxial CATV networks and copper pair telephone networks are generally unable to handle the tremendous amounts of electronic data that must be transmitted, should the availability of interactive services become universal.

One solution to overcome the limitations introduced by the physical networks is to replace the current CATV and telephone networks with fiber optic networks, which are capable of carrying much more information. However, total replacement of an established network with fiber optic components is cost prohibitive. Although the use and diversity of interactive services is expected to grow in the future, the extent of the growth is unknown. Additionally, there are many types of services that have not yet been envisioned but will require support from any network that is established.

Accordingly, there exists a need for an information transport infrastructure that is capable of efficiently and optimally transporting digital and analog communications between information service providers and service requestors. This infrastructure must be capable of supporting existing services as well as expanding to support future services.

The network architecture of the present invention allows distributed delivery of broadcast and interactive digital services over a hybrid fiber-coax metropolitan area network to service requestors. The network provides an information transport infrastructure that increase the capability of hybrid fiber-coax transmission systems. The network includes at least one remote/local hub that communicates with a plurality of video information providers (VIPs) and a plurality of headends/central offices. Each headend communicates with a plurality of settop terminals. The hub includes a backbone subnetwork which provides the physical medium for components within the hub to communicate with each other, and for entities located outside the hub, such as the VIPs and the headends, to communicate with, and through, the hub. The hub processes broadcast digital information services from the VIP and distributes the services directly to the plurality settop terminals. The headends facilitate interactive communications between the VIPs and the settop terminals served by that particular headend.

The network architecture permits two-way transparent (protocol stack independence, layer 3-7) data transport service between the VIPs and the video information users (VIUs) at the STTs. Through frequency division multiplexing, the network architecture facilitates analog signal distribution. Accordingly, the network is compatible with distribution systems which provide analog services.

Accordingly, it is an object of the present invention to provide network architecture for the distributed delivery of broadcast and interactive digital services.

It is a further object of the invention to provide a network architecture which is compatible with distribution systems capable of providing analog services.

It is a further object of the invention to provide a network architecture that provides for equitable access by multiple video information providers.

It is a further object of the invention to provide a network architecture that does not impose any protocol stack requirements on the communications from the video information providers.

It is a further object of the invention to provide a modular and scalable network architecture which allows for incremental agreement deployment by the network provider.

It is a further object of the invention to provide a network architecture that is easily tailored to and is independent of the type of network provider (a cable operator or a regional phone company).

Other objects and advantages of the system will become apparent to those skilled in the art after reading the detailed description of a presently preferred embodiment.

BRIEF DESCRIPTION OF THE DRAWINGS

Figure 1 is a block diagram of an end-to-end cable television communication network embodying the present invention;

Figure 2 is a block diagram of the remote/local hub of the present invention;

Figure 3 is a block diagram of a headend/central office of the present invention;

Figure 4 is a block diagram of the cable television distribution network of the present invention;

Figure 5 is a block diagram of a settop terminal of the present invention;

Figure 6 is a prior art layered protocol stack;

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Figure 7 is a prior art link level frame format used in conjunction with the protocol stack of Figure 6;

Figure 8 is the ISO-OSI based protocol stack during protocol syntax processor upgrade in accordance with the present invention;

Figure 9 is an example of a frame format of the link layer protocol;

Figure 10 is a flow diagram of the migration from an old protocol syntax processor to a new protocol syntax processor;

Figure 11 is the frame format of Figure 9 including the test frame structure;

Figure 12 is a flow diagram of the testing procedure of a new protocol syntax processor;

Figure 13 is an example of values transmitted in the test frame format for a field check;

Figure 14 is a flow diagram of the recovery procedure of the slave node of the present invention;

Figure 15 is a time division multiple access frame utilized in the preferred embodiment of the hybrid MAC;

Figure 16 is a block diagram of the space, frequency and time domains of the hybrid MAC;

Figures 17A and 17A-1 are a flow diagram of the (RSR/ASA) time division multiple access component of the hybrid MAC;

Figure 17B is a flow diagram of the multi-rate dynamic time diversion multiple access component of the hybrid MAC;

Figure 18 is the preferred DLL packet sublayer of the present invention; and

Figure 19 is the preferred MPEG2 based DLL message sublayer of the present invention.

TABLE OF ACRONYMS

5	AAL5	ATM Adaption Level 5
	AC	Addressable Controller
	ASEM	Access Subnetwork Element Manager
10	ATM	Asynchronous Transfer Mode
	APP	Adaptive Protocol Processor
15	CLP	Cell Loss Priority (ATM)
	CRC	Cyclic Redundancy Code
	DLL	Data Link Layer
20	DSA	Dynamic Slot Allocation
	DTE	Data Terminal Equipment
	ECM	Entitlement Control Messages
25	EIA	Electronics International Association
	EMM	Entitlement Management Messages
	FDM	Frequency Division Multiplexing
30	FTTN	Fiber To The Node
	GFC	Generic Flow Control (ATM)
35	HEC	Head Error Check (ATM)
	HFC	Hybrid Fiber Coax
	IBTM	In-band Transport Multiplex
40	IP	Internet Protocol
	IR	Infrared
	IPPV	Impulse Pay Per View
45	ITEM	Integrated Transport Encryption Multiplexer
	LAN	Local Area Network
	LLC	Logical Link Control
50	L1G	Level One Gateway (regulated by the FCC)
	L2G	Level Two Gateway (unregulated by the FCC)

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	MAC	Medium Access Control
	MPEG2	Motion Picture Expert Group 2
5	MPTS	Multi Program Transport Multiplex (MPEG2)
	NVRAM	Non-Volatile Random Access Memory
10	PES	Packetized Elementary Stream
	PSI	Program Specific Information
	OAM&P	Operation Administration Maintenance and
15		Provisioning
	OBTM	Out of band transport multiplex
	OSI	Open Systems Interconnection
20	OSS	Operation Support System
	PAT	Program Association Table (MPEG2)
25	PCR	Program Clock Reference (MPEG2)
	PCS	Personal Communication Services
	PDU	Protocol Data Unit
30	PID	Packet Identifier
	PMT	Program Map Table
	POP	Plain Old Polling
35	POTS	Plain Old Telephony Service
	PPV	Pay Per View
	PSI	Program Specific Information
40	PSP	Protocol Syntax Processor
	PT	Payload Type
45	QAM	Quadrature Amplitude Modulation
	QPSK	Quadrature Phase Shift Keying
	RSR	Random Slot Reservation
50	SDU	Service Data Unit
	SPTM	Single Program Transport Multiplex (MPEG2)

STT	Settop Terminal
VCC	Virtual Channel Connection (ATM)
VCI	Virtual Channel Identifier (ATM)
VDT	Video Dial Tone
VIP	Video Information Provider (information owner)
VPI	Virtual Path Identifier (ATM)
WAN	Wide Area Network

A CATV communication network 10 embodying the present invention is shown in Figure 1. The communication network 10 generally comprises a remote/local hub 14 which communicates with a plurality of headends/central office 18, each of which in turn communicate with a plurality of settop terminals (STTs) 16. The STTs 16 are the interface between the television of a video information user (VIU) and the communication network 10. The remote/local hub 14 may be physically located remote from the headends 18 or, alternatively, may be located at the site of any one of the headends 18. The communication network 10 interfaces with a plurality of video information providers (VIPs) 12 which provide compressed digital video and services. Through the remote hub 14 and the headends 18, the communication network 10 provides two-way transparent (protocol stack independence, layer 3-7) data transport service between the VIPs 12 and the video information users at the STTs 16. The hub 14 provides broadcast information services from the VIPs 12 to all STTs 16 on the network 10. The headends 18 facilitate interactive communications between the VIPs 12 and the STTs 16 that are served by that particular headend 18. In the preferred embodiment of the invention, communications between the VIPs 12, the remote/local hub 14 and the headend/central offices 18 are transmitted over a fiber optic medium.

To provide the bi-directional communication flow over the network 10, frequency spectrum of the physical medium from the headend 18 to the STTs 16 is divided into a downstream signal path originating at the headend 18 and an upstream signal path originating at the STTs 16. The bandwidth of the physical medium in the preferred embodiment extends up to 1 GHz. The downstream bandwidth typically employs frequencies above 50 MHz, and the upstream frequencies below 50 MHz. The downstream and upstream bandwidths are further divided into 6 MHz channels. In the present invention, a portion of the 6 MHz channels are allocated for analog communications and the remainder for digital communications. Accordingly, analog and digital communications may be frequency division multiplexed (FDM) over the separate channels and transported over the same physical medium. Analog CATV communication systems are well known in the art, such as the system disclosed in U.S. Patent No. 4,533,948, (to McNamara et al.) and in U.S. Patent No. 4,245,245, (to Matsomoto et al.), which are incorporated by reference as if fully set forth.

Referring to Figure 2, a remote/local hub 14 made in accordance with the teachings of the present invention is shown. The hub 14 includes a level 1 gateway (L1G) 20, an access subnetwork element manager (ASEM) 22, an addressable controller 24, a billing system 26 (co-located or remotely located), an operations support system (OSS) 28 (co-located or remotely located), an integrated transport encryption multiplexer (ITEM) 30, a 64 quadrature amplitude modulation (QAM) modulator 32, an RF upconverter 34, a quadrature phase shift keying (QPSK) modulator(s) 36 (optional) and an asynchronous transfer mode (ATM) services multiplexer 38. A backbone subnetwork 40 provides the physical medium and the optical to electrical interface for components within the hub 14 to communicate with each other and for outside entities (such as the VIPs 12 and STTs 16) to communicate with, and through, the hub 14. Communications between the hub 14 and the headend 18 elements are conducted via Internet protocol, asynchronous transfer mode (IP/ATM) signaling in WAN connectivity and IP/Ethernet in LAN connectivity. There may be more than one of each of these components depending upon system capacities (i.e. desired number of channels for the ITEM 30, number of subscribers for the AC24 etc.).

The specific components which comprise the network architecture of the present invention will now be presented in detail. The video information provider (VIP) 12 consists of a level two gateway (L2G) and associated servers. The L2G acts as an interface between the VIP 12 and the network 10. The VIPs 12 are the source of live, archival broadcast or interactive service content, (comprising electronic encyclopedias, electronic catalogs, downloadable applications, movies, etc.), communications with the network 10, and service menus. The L2G communicates with the L1G 20 to manage the establishment and termination of service/session connections between the VIUs and the VIPs 12. The L2G

also provides menu presentation and selection processing to and from the VIUs, performs VIU authentication and forwards billing information, for VIP 12 provided services to the billing system 26.

The level 1 gateway (L1G) 20 provides overall management of network 10 in support of service delivery from a VIP to a VIU(s). The L1G 20 performs the following functions: 1) management of VIP-VIU interactive sessions through communication with the VIPs 12; 2) VIU authentication and collection of billing information relating to network 10 support of VIP 12/L1G 20 provided services for forwarding to the billing system 26; 3) interactive menu presentation which is responsive to the service selected; and 4) database management of VIU profiles for digital broadcast and interactive services.

The access subnetwork element manager (ASEM) 22 acts as an agent to (i.e. is controlled by) the L1G 20 and/or OSS (depending on the functions to be performed by the ASEM 22). At the direction of the L1G 20, the ASEM 22 provides management of headend 18 components (shown in Figure 3), much in the same manner as the L1G 20 oversees management of resources on the backbone subnetwork 40 (via a designated backbone network manager not shown). The ASEM determine which item and NC can accommodate a new connection (based on already utilized resources within each) and the L1G 20. The ASEM 22 conveys parameters such as ATM virtual path identifier (VPI) and virtual channel identifier (VCI) values for session signaling and session content to the ITEMS 30, 50 and the network controllers 62. The ASEM 22 also conveys associated transmission rates for the downstream ATM VPI/VCI and signaling rates for upstream as conveyed to it by the L1G 20 (these values are originated by the VIP 12/L2G). The ASEM 22 forwards appropriate scheduling parameters to the addressable controller 24 for encryption of pay-per-view (PPV) and impulse-pay-per-view (IPPV) services by the ITEMS 50. In the preferred embodiment of the invention, the ASEM 22 oversees OAM&P (Operation Administration Maintenance and Provisioning) functions through the backbone subnetwork 66 and interfaces with the OSS center 28 to convey status or any OAM&P information of elements within the hubs (Mux/Mods, Demod/Muxes, etc.) to the OSS 28.

The addressable controller 24 manages the secure delivery of broadcast, pay-per-view and non-video-on-demand (staggercast) services, including VIU authorization for those services and program scheduling by controlling the encryption subsystem. Access control and encryption parameters are forwarded to network elements which perform downstream and upstream encryption and decryption. In the preferred embodiment of the invention, downstream encryption is implemented in the ITEMS 30, 50 and downstream decryption is implemented in network modules 70, which are part of each STT 16. Upstream encryption is implemented in the network module 70 and upstream decryption is performed by a network controller 62. For interactive service communications, which are facilitated at the headends 18, the addressable controller 24 preprovisions the ITEMS 50 and the network modules 70 with the appropriate encryption/decryption parameters. For broadcast service communications, which are facilitated by the hub 14, the addressable controller 24 forwards similar parameters to the ITEM 30 based on scheduling information forwarded by the L1G 20 through the ASEM 22.

The integrated transport encryption multiplexer (ITEM) 30 provides secure delivery of broadcast digital services information to the VIUs as an in-band transport multiplex (IBTM). The ITEM 30 originates the IBTM including video, audio, data by performing ATM to MPEG2 reassembly and readaption (AAL5) of single program transport multiplex (SPTM). This includes ATM to MPEG2 reassembly of audio-visual content, ATM to AAL5-service data units (SDU) reassembly of non-MPEG2 data and removing jitter and adjusting program clock reference (PCR) timing for audio visual content. The ITEM 30 creates an aggregate MPEG2 multiplex from any input SPTM to any output multi-program transport multiplex (MPTM). In doing so, the ITEM 30 uniquely reassigns packet identifier (PID) values, creates an aggregate stream (program specific information (PSI) including a program association table (PAT) and a program (PMT) map table and selectively encrypts SPTMS as instructed by the ASEM22. The ITEM 30 updates the aggregate stream PCR and inserts entitlement control messages (ECMs). It also performs similar operations when multiplexing L1G 20 signaling and addressable controller 24 messages (including entitlement management message (EMMs)) on the out-of-band transport multiplex (OBTM) which is then forwarded to the QPSK modulator 36.

The ITEM 30 provides a transport interface for digital broadcast communications between the VIPs 12 and the STT 16 via the backbone subnetwork 40 (or through direct interfaces if co-located). Broadcast audiovisual information and signaling originated by the VIPs 12 is forwarded to the STT 16 via the ITEM 30. More specifically, the ITEM 30 accepts an ATM stream via an ATM/SONET interface (with sustained cell rate version of AAL5) and selects the appropriate cells for MPEG2 packet reassembly based on the value of the VCI which is conveyed by the ASEM 22 to the ITEM 30 during connection establishment. This is achievable since the virtual connection between the VIP 12 and the VIU is maintained end-to-end. The resulting MPEG2 transport multiplex (consisting of multiple audiovisual information streams, protocol-independent, from an ITEM 30 standpoint, information streams carried as AAL5-SDUs (such as stock quotes) is output to the 64 QAM modulator 32.

In order to ensure secure delivery of a given broadcast audiovisual service, the ITEM 30 selectively encrypts, pursuant to the addressable controller 24 configuration, a given set of MPEG2 program streams within the MPEG2 transport stream. Access control and encryption related information is forwarded to ITEM 30 from the addressable controller 24. The ITEM 30 incorporates this information within the MPEG2 transport stream (per MPEG2 rules where applicable) and encrypts as instructed by the addressable controller 24. Each output of the ITEM 30 is forwarded to a single 64

QAM modulator 32 whose output is converted to the selected downstream channel by an RF upconverter 34 and then combined by an RF combiner 35 for downstream transmission. Each ITEM 30 keeps track of the resources required for a given service.

The QAM modulator 32 accepts the downstream IBTM output of ITEM 30, applies forward error correction and 64 QAM encoding. The output of the QAM modulator 32 is input to the RF upconverter 34.

Downstream OBTM signaling intended for all subscribers from the L1G 20 or the addressable controller 24, (and optional application downloads from the L1G 20, or L1G 20 menus to VIPs 12), are sent from the ITEM 30 to the QPSK modulator 36. The QPSK modulator 36 accepts the output of the ITEM 30. The QPSK mod 36 also provides forward error correction and QPSK modulation for the OBTM transmission. The output is input to the RF combiner 35 for downstream transmission.

The RF upconverter 34 receives the outputs from the QAM modulator 32 for IBTM (and the QPSK modulator 36 for OBTM if upconversion was not performed by the QPSK mod 36). Each RF upconverter 34 is capable of accepting the output from two modulators 32, 36. The RF upconverter 34 accepts two intermediate carrier frequencies (41.0 - 47.0) with an associated RF output which is frequency agile from 50 to 1000 MHZ. The upconverter 34 will accept either the QAM modulator output 32 at IF from the ITEM 30 or an analog television channel input (not shown) at IF. The analog inputs will be upconverted to predetermined analog channels. The RF upconverter 34 is controlled either via front panel switches or remotely via the Ethernet interface. The output from the RF upconverter 34 is input to an RF combiner 35.

The RF combiner 35 is a full bandwidth 5 to 1000 MHZ 12-way combiner/splitter. It may be used to merge signals from up to twelve channels or to distribute a single signal to twelve outputs. The RF combiner 35 employs directional coupler circuitry to attain high channel isolation for protection from channel to channel interference. The output from the combiner 35 bypasses the processing within the headend 18 and proceeds directly to the headend combiners (not shown) for further combination with the output from the headend 18.

The asynchronous transfer mode (ATM) services multiplexer 38 aggregates the non-null ATM cells from each DS3 interface (up to n DS3 interfaces, where n ranges between 1 and 18 per ATM services mux 38) and forwards the multiplexed cells to the backbone network 40. It also acts as an ATM edge device for the OAM&P in the headend 18 providing virtual LAN connectivity between the hub 14 and the headends 18. The backbone subnetwork 40 interconnects the LAN 39 where the ASEM 22 and the addressable controller 24 reside to the LAN 61 within the headend 18 to appear as though all components reside on the same LAN.

The operation support system 28 provides the OAM&P services in support of the network 10. The billing system 26 stores the billing information related to each VIU and generates billing statements. The general operation and functions of both of these components is well known to those skilled in the art.

Referring to Figure 3, a headend 18 made in accordance with the teachings of the present invention is shown. The headend 18 facilitates digital interactive communications between the VIPs 12 and the VIUs. All of the digital broadcast communications which were processed by the hub 14 are further combined at the output of the headend 18 by combiners (not shown). Accordingly, no further processing of the broadcast communications by the headend 18 is required.

The headend 18 includes an ITEM 50, a 64 QAM multiplexer/modulator (mux/mod) 52, an RF upconverter 54, a QPSK multiplexer/modulator (mux/mod) 58, a QPSK demodulator/multiplexer (demod/mux) 60, a network controller 62 and a headend ATM services mux 64. The headend 18 communicates with a plurality of STTs 16 through the CATV transmission network 56. Each STT 16 includes a network module 70 for interfacing with the CATV transmission network 56.

The ITEM 50 provides secure delivery of interactive digital services information to the VIUs as an IBTM. Additionally, it can be configured to provide broadcast services to subscribers served by that particular headend 18. The ITEM 50 originates the IBTM including video, audio, data and VIP 12 signaling by performing ATM to MPEG2 reassembly and re-adaption (AAL5) of SPTM's. This includes ATM to MPEG2 reassembly of audio-visual content, ATM to AAL5-SDU's reassembly of session signaling or non-audio-visual session content, and removing jitter and adjusting PCR timing. The ITEM 50 creates an aggregate MPEG2 multiplex, (from the content and signaling of multiple sessions,) from any input SPTM to any output MPTM. As is performed by the ITEM 30 in the hub 14, the ITEM 50 uniquely reassigns PID values, creates an aggregate stream PSI and encrypts each SPTM. The ITEM 50 also updates the aggregate stream PCR and inserts ECMs. It also performs creates an aggregate multiplex of L1G 20 signaling and addressable controller 24 messages, including EMMs, on the OBTM which is then forwarded to the QPSK mux/mod 58. Each ITEM 50 keeps track of the resources required for the active sessions. Should the ASEM 22 need to recover its database, it queries the different ITEMS 50 it oversees through the OAM&P interface (SNMPv2/UDP/Ethernet).

The single physical interface (OC-3c) into the ITEM 50 from the backbone subnetwork 40 allows for rate policing of both session content and signaling by the backbone subnetwork 40.

The QAM multiplexer/modulator 52 accepts the downstream IBTM output of ITEM 50, adds signaling information in support of medium access control (MAC), inserts medium-access control synchronization information and applies forward error correction and 64 QAM encoding. The output of the QAM mux/mod 52 is input to the RF upconverter 54.

Downstream OBTM signaling from the L1G 20 or the addressable controller 24, (and optional application downloads from the L1G 20, or L1G 20 menus to VIPs 12), are sent from the ITEM 50 to the QPSK multiplexer/modulator

58. The QPSK mux/mod 58 accepts the output of the ITEM 50 and multiplexes MAC information as in the in-band case. The QPSK mux/mod 58 also provides forward error correction and QPSK modulation for the OBTM transmission. The output is input to the RF combiner 59 for downstream transmission.

The primary function of the network controller 62 is to administer medium access control (MAC) functions for the network module 70 interfacing to the network 10. MAC is a sublayer of the data link layer which coordinates the transmission of data from various network modules 70 to the headend 18 based on fair access and predictability of performance. The network controller 62 determines the MAC operating parameters based on those supplied by the ASEM 22 including but not exclusive to: 1) fiber to the node (FTTN) size; 2) upstream spectrum allocation (8-12MHz, 8-15MHz); 3) return path transmission rates; and 4) QPSK mux/mod 58 and QPSK demux/mod 60 configuration (i.e. connectivity to the network controller 62 which will allow the network controller 62 to route information such as acknowledgments (to upstream transmitted packets), through the appropriate QPSK mux/mod 58 or QAM mux/mod 52.

The network controller 62 also administers the type of upstream access that will be required: 1) plain old polling (POP); 2) default assigned TDMA; and 3) dynamically assigned TDMA carriers (frequency and time slot assignments). The resources allocated by the network controller 62 for connection-oriented MAC service requests are based upon the desired session bandwidth passed by the ASEM 22 on behalf of the L1G 20 or the VIP 12. Medium access control acknowledgment messages and information are forwarded to the STTs 16 over Ethernet via the QAM mux/mod 52 and the QPSK mux/mod 58.

The network controller 62 supports upstream access for interactive services requested from STTs 16 by: 1) administering the adaptive MAC operation to maintain guaranteed minimal upstream latency and ensure fair upstream access by all subscribers; and 2) forwarding upstream data to the L1G 20, L2G, or addressable controller 24 via the ATM services mux 64. The network controller 62 also allows for the dynamic provisioning of multiple effective transmission rates according to the needs of the applications and services utilizing the network 10.

The network controller 62 alleviates the L1G 20 from any functions pertaining to upstream access by overseeing network module 70 initialization (default carrier, default time division multiple access (TDMA) transmission time slots, etc.) as well as overseeing dynamic TDMA carrier and time slot assignment whenever a VIP-VIU session is established.

In the preferred embodiment, the network controller 62 is configured to accommodate $n \times (1.5 \text{ Mbps})$ streams from a QPSK demod/mux 60 ($1 < n < 5$) and forwards the upstream ATM cells to an appropriate external routing/switching element.

The network module 70 interfaces the STT 16 to the hybrid fiber coaxial (HFC) physical medium 56. The network module 70 facilitates the extraction of RF signals and demodulating (64QAM or QPSK) these signals. Forward error correction is then performed before any data link layer (DLL) processing takes place. As part of the DLL processing, the network module 70 may decrypt certain IBTM or OBTM components and perform AAL5-SDU based (upper layer) signaling extraction. It forwards AAL5-SDU protocol data units (PDU) to the STT central processing unit as well as the service stream to the other STT processing elements.

The network module 70, under the management of the network controller 62 forwards signaling from the STT 16 to the corresponding network controller 62 which forwards this information to the L1G 20 or to the VIP 12 through the headend backbone subnetwork 66 and the hub backbone subnetwork 40. The network module 70 also communicates with the addressable controller 24 for access control and decryption/encryption authorization.

The QPSK demod/mux 60 receives up to six upstream carriers, demodulates the carriers and performs forward error correction. The resulting ATM cell streams are multiplexed to form a nominal 1.5 Mbps stream which is forwarded to the network controller 62. Additionally, the QPSK demod/mux 60 forwards measured values of certain physical layer parameters to the network controller 62 for network module 70 power calibration, and performs upstream synchronization supporting functions to facilitate ranging of STTs 16 (propagation delay calibration) for optimal TDMA operation.

The components shown in Figure 3 which have not been accompanied herein by a specific description operate as the equivalent components shown in Figure 2.

Although the aforementioned description of the specific components provides an understanding of the function of each component, a thorough understanding of the network architecture of the present invention will be further facilitated by a description of the signaling and communications between network components.

The information from a VIP 12 to a STT 16 flows on an IBTM which is modulated as a 64QAM signal, while information from the L1G 20 to a STT 16 flows through either an IBTM or an OBTM, modulated as a QPSK signal. A downstream path consists of both an IBTM and an OBTM destined to a STT 16. An upstream signaling path consists of STT 16 to L1G 20, VIP 12, addressable controller 24, or network controller 62 signaling, via the network module 70, through the network controller 62, through the backbone subnetwork 40. In addition, an upstream path consists of signaling between the network module 70 and the network controller 62. All upstream signals are QPSK modulated.

With respect to the downstream path IBTM, the CATV communication network 10 uses the backbone subnetwork 40 to interconnect the VIP 12 to the ITEMS 30, 50. Digital streams containing compressed video material embedded in ATM cells originate at the VIP 12. AAL5 adaptation is used to transport a MPEG2 SPTM over ATM. Additionally, signaling originating from the VIP 12 to the STT 16 is carried as IP/ATM (or any protocol/ATM) using AAL5 adaptation as well. The ITEMS 30, 50 accept a plurality of the ATM streams, as instructed by the ASEM 22, to create an aggregate

MPTM whose aggregate nominal rate is 27Mbps. The VIP 12 informs the L1G 20 of the service and signaling rates required in support of a given SPTM. The L1G 20 in turn forwards this information to the ASEM 22. The ASEM 22 determines which of the ITEMS 30, 50 present within a given headend 18 can accommodate this SPTM. The ASEM 22 then conveys this parameter along with other parameters such as MPEG2 program number to be assigned to the SPTM, ITEM 30, 50 output port (1 of 6). This mapping from ATM VCI to MPEG2 program number allows the reservation of the virtual channel (connection) end-to-end through the ITEM 30, 50 (and terminated at the STT 16) whenever a new SPTM has to be multiplexed for delivery to a single SST 16 or multiple STTs 16, (single STT 16 in the case of an interactive service and multiple STTs 16 in the case of a broadcast service). The output of the ITEM 30, 50 consisting of a MPTM is then forwarded to the 64 QAM mux/mod 32, 52 and then to the RF upconverter 34, 54 to place the 64QAM signal in the appropriate 6MHz channel.

With respect to the downstream path OBTM, the output of the L1G 20 signaling is delivered to the ITEM 30, 50 through the same OC-3c interface as the IBTM. Each ITEM 30, 50 has the capability of generating an OBTM, carrying L1G 20 signaling for L1G-VIU default connections as well as EMMs originating from the addressable controller 24. The OBTM output of the ITEM 30, 50 is forwarded to a QPSK mux/mod 36, 58.

With respect to the upstream path, the STT 16 central processing unit forwards signaling to the network module 70 which segments the data into individual ATM cells (AAL5) and forwards the cell based PDUs to the STT's 16 QPSK modulator (not shown). Any non-VIP-VIU session signaling is sent over a default upstream carrier (such as that destined to the L1G 20 or network controller 62). All session-related signaling is forwarded on a dynamically assigned carrier which is assigned by the network controller 62 during session connection establishment. The VIP 12 specifies the effective session upstream signaling rate and the downstream session signaling and content rate, depending on the activated application, to the L1G 20. The L1G 20 in turn forwards these parameters to the ASEM 22. The downstream rates are forwarded to the ITEM 30, 50 while the upstream signaling rate is provided to the network controller 62. A default upstream signaling rate is assumed by the network controller 62 if the VIP 12 leaves this parameter unspecified. The ASEM 22 also forwards the upstream VPI/VCI pair to the network controller 62 on VIU-VIP session by session basis, which in turn informs the network module 70 during connection establishment phase. The network module 70, as part of the initialization process, is aware of the default upstream carrier and the VPI/VCI values primarily used for STT-L1G signaling.

Each upstream carrier is demodulated by a QPSK demod/mux 60 which, in addition to demodulating a given number of carriers, multiplexes the individual upstream PDUs (preserving PDU boundary) and forwards the multiplex to the network controller 62. The network controller 62 examines each PDU and processes it by either forwarding it to the ATM services mux 64 (which performs an ATM relay function to the L1G 20 or the VIPs 12) or forwarding it to the appropriate processing element within the network controller 62 when the PDUs are MAC specific.

Referring to Figure 4, the topology of the signal transmission network 56 between the headend 18 and the STTs 16 is preferably a hybrid star followed by a tree and branch topology. In the predominantly star-type aspect of the hybrid fiber-coax network, a fiber-optic line 90 is provided from the headend 18 to each node 92. Each node 92 includes forward and return signal amplifiers and splitters/combiners to support the bi-directional communications. The single optical input 90 is converted into an electrical signal which is split into four coaxial feeder lines 94 which are generally dedicated to a limited group of STTs 16 such as neighborhood. In the predominantly tree and branch aspect of the hybrid network, the feeder lines 94 connect the node 92 to the coaxial drop lines 96 of individual STTs 16. At points where the cable divides, signal splitters/combiners are installed. Signal amplifiers 98 may also be provided along the coaxial feeder lines 94 as required to boost the RF signals and ensure that nominal signal strength levels are maintained.

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Table 1 provides the I/O interfaces of each component in the preferred embodiment.

TABLE 1

COMPONENT	INTERFACES
Addressable Controller	I/O: - 2/10BaseT Ethernet - 32 x RS-232 (4 x multipin-connector driving 8 port concentrator)
Network Controller (NC 1000)	Input: - 5 x EIA/RS-485 DTE I/F
	Outputs: - 1 x ATM/DS3 (information rate <=9 Mbps)
	I/O: - 2 x Ethernet 10BaseT
RF Modules Data Interfaces	64QAM MUX/MOD - Input: 1 x TAXI @ 27Mbps - Output: 1 x IF @ 43.75 MHZ (75 Ohm F-Connector) - I/O: 2 x Ethernet 10BaseT
	QPSK MUX/MOD: - Input: 1 x EIA/RS-530 @ 1.5Mbps - Output: 1 x RF, Range: 71-129 MHZ (75 Ohm F-Connector) - I/O: 2 x Ethernet 10BaseT
	QPSK DEMOD/MUX: - Input: Upto 6 RF Inputs - Output: 1 x EIA/RS-485 @ 1.5 Mbps - I/O: 2 x Ethernet 10Base T
Integrated Transport Encryption Mux (ITEM 1000) Data Interfaces	Input: - 1 x Optical Carrier 3-Concatenated (OC-3c, 155.52 Mbps)
	Outputs: - 5 x TAXI @ 27 Mbps - 1 x RS-530 @ 1.544 Mbps
	I/O: - 1 x Ethernet 10Base T - 1 x RS-232 @ 19.2 Kbps
Network Module Interfaces	Input: - RF via F Connector

COMPONENT	INTERFACES
	Output: - RF Bypass via F Connector Video via RCA phono plug Audio Right and Left Channels via RCA phono plug
	I/O: - Digital bidirectional interface, (32-pin molex connector)

Referring to **Figure 5**, a SST 16 is shown. Communications in the downstream path originating at the headend 18 are transmitted through the coaxial drop line 96. Video information signals are processed through a frequency agile tuner 110 and a descrambler timing and control module (optional - if analog video is scrambled) 112 before entering a television set 114. The tuner 110 is responsive to the frequency of the downstream channel selected by the VIU to remove the carrier signal. The descrambler 112 descrambles the baseband signal of the selected channel if the VIU is an authorized user. Similarly, digital video passes through the decryption and then MPEG2 decodes and D/A conversion to forward the composite video for display. (The baseband video signal is placed on a second carrier signal frequency, typically television channel 3 or 4, for input into the television 114). The tuner 110 and descrambler 112 are controlled by the network module 70, which includes an encryption/decryption controller 124 and a MAC module 126.

The network module 70 interfaces with a processor 116 which comprises a central processing unit (CPU) 118, a read only memory (ROM) 120 and a non-volatile random access memory (RAM) 122. Several processing components reside in the RAM 122, including an existing protocol syntax processor 128 (PSP), an adaptive protocol processor 130 (APP), a memory manager 132 and a transmit queue 134. The remaining portion of the RAM 122 is available for processing higher layers of the protocol and for general use. The APP 130, which includes a primitive PSP, is also resident in ROM 120 for initial start-up and recovery operation. The CPU 118, in conjunction with the different processing components in RAM 122, allows the STT 16 to read incoming data frames, parse the frames by sequentially stripping off each nested layer, and perform the application embedded therein.

Data frames within the OBTM forwarded through the headend 18 to the STT 16 are received through a second frequency agile receiver 140, which is tuned to the particular control channel for receiving control messages. In the preferred embodiment, the transmitter 142 transmits communications upstream from the STT 16 to the network controller 62.

The STT 16 is controlled via the subscriber remote control 144. The remote control 144 has an infrared (IR) signal emitter 146 which sends IR control signals to the IR receiver 148 as is well known in the art. The received control signals are then forwarded to the processor 116.

In the preferred embodiment of the present invention, the network architecture facilitates the upgrading of STTs 16 by permitting remote replacement of the PSP 128 resident within the STT 16 with a new PSP 129 without interrupting operation of the network 10 or the STT 16.

Figures 6-14 illustrate an adaptive protocol wherein a protocol syntax processor resident within a SST 16 may be remotely upgraded. This will be described in detail in what follows.

Figure 6 illustrates the basic layered protocol model which is well known to skilled artisans. The frame 150 comprises a signal transmitted through the network 10 from a source node, such as the network controller 62, to a destination node, such as the STT 16.

The frame 150 is arranged in nested layers. The layered model of **Figure 6** closely conforms to the reference model for Open Systems Interconnection (OSI) developed by the International Standards Organization, Geneva, Switzerland. Specifically, the ultimate "data to be transferred" is in the application layer 152. The application layer 152 is nested within the presentation layer 154. The presentation layer 154 is nested within the session layer 156, which is nested within the transport layer 158, and so on through the network layer 160 and the link layer 162. In general, each layer includes a header 164 (which precedes the application data 152) and a trailer 168 (which follows the application data 152). However, for any given layer, the respective header may be minimal and the respective trailer may be nonexistent.

The source node e.g. VIP 12, L1G 20, AC 24, NC 62 assembles the frame message layers (3-7) 150 before transmission to the STT 16 (destination node). The destination node parses the frame 150 one layer at a time. First, the destination node strips the link layer 162 from the frame. The "data" of the link layer 162 is the network layer 160. The "data"

of the network layer 160 is the transport layer 158, and so on down to the application layer 152. Each layer of the protocol is assigned specific system communication functions as briefly outlined below.

The physical layer (not shown) is concerned with the electrical and mechanical characteristics associated with the physical network termination plug or socket. This includes the type of electrical signals used, the size and configuration of the connecting I/O plugs and the number and arrangement of pins.

The link layer 162 provides the means to transfer data reliably over the medium. As detailed above, the link layer 162 performs basic communication functions of medium access control when a shared medium is used, generation and checking of error detection codes, and source and destination node addressing within the network 10. The link layer 162 for shared media is typically divided into two sub-layers: the MAC sub-layer and the logical link control (LLC) sub-layer.

The network layer 160 specifies addressing and routing instructions to establish connections between the source and destination nodes located in different networks. The network layer 160 enables two networks to interconnect across one or more sub-networks, thereby providing a uniform end-to-end service to the transport layer.

The transport layer 158 governs the data flow between the source and destination nodes once a signal path connection has been established. Typical functions of the transport layer 158 include data flow control, error control and message fragmentation.

The session layer 156 provides the means for two communicating nodes to control their subsequent communication. It also provides means to allow a communication to be restarted at a pre-determined synchronization point in the event of a communication failure. The session layer 156 comprises a set of functional units which must be established when the session connection is set up.

The presentation layer 154 controls the format and syntax of data that is being exchanged between two communicating nodes. After a connection has been established, the presentation layer 154 allows the communicating nodes to establish a common syntax for the data to be exchanged.

The application layer 154 includes the programs to be utilized by the subscriber. For example, in a video-on-demand application, the application layer 154 may contain specific instructions for L2G menu navigation and selection by authorized subscribers.

Figure 7 shows the prior art frame format 180 for the link layer protocol. A frame 180 includes a synchronization field 182 a protocol version field 184, a destination address field 186, a source address field 187, a control field 188, a link level message field 190 and a cyclic redundancy code field (CRC) 191.

In contrast, Figure 8 illustrates an ISO-OSI based layered protocol stack during protocol syntax processor upgrade. For clarity, only the heading of each layer is shown. The heading and the tail may include a full message, a nominal message or no message at all depending upon the frame to be sent. A given protocol stack may be comprised of any combination of layers. Further, not all layers need to be present in each protocol stack. The source node assembles the frame message starting with the application layer 192. The presentation layer 194 is then added, followed by the session layer 196, the transport layer 198 and the network layer 200. The protocol stack includes two data link layers: a new data link layer 202 which is nested within the old data link layer 204.

Figure 9 is an example of frame format of the link layer protocol 206. The frame 206 includes an eight bit synchronization field 208, an eight bit revision field 210, a twenty-four bit destination address field 212, a twenty-four bit source address field 213, an eight bit control field 214, an eight bit control subtype field 215, an eight bit length field 216, a link level message field 217 which may vary in length, and an eight bit CRC field 225.

The protocol sent from the master node to the slave node contains instructions regarding the PSP 128 for the APP 130 to follow. The control field 214 within the link layer protocol 206 identifies the type and format of the information to follow in the control subtype field 215, the length field 216, and the link level message field 217. The control field 214 prompts the APP 130 to receive and process the frames within the protocol in a certain manner. As shown in Table 2, the APP 130 may be instructed to update the existing PSP 128, accept a completely new PSP 129, or perform other actions on the PSP 128. It should be evident to those skilled in the art that other control field types may be specified at a later date and included within a new or revised protocol syntax processor. Depending upon the type of control field 214, additional fields may be necessary. The number and length of additional fields are specified within the control subtype 215 and length 216 fields, respectively. The additional fields are then transmitted within the link level message field 217. The length field 216 indicates the length of the link level message 217 which, in essence, extends the length of the header. The length field 216 is indicative of how far the APP 130 must parse the frame in order to get to the new protocol. In this manner the standard length of each field, the number of fields and the function of each field may be changed without physically replacing the node.

TABLE 2

	CTRL FIELD TYPE	LENGTH FIELD	ADDITIONAL FIELDS
0	Reserved	-	-
1	Update PSP	-	-
2	New PSP	X	sequence number image section
3	Last PSP	X	sequence number image section
4	Test New PSP	X	subtype, number of fields,...
5	Stand By	-	-
6	Activate PSP	-	-
7	Set New To Current	-	-
8	Use Current PSP	-	-

The procedure for replacing a node's PSP 128 is shown in Figure 10. Migration from the old PSP 128 to a new PSP 129 is controlled by a "master" node which directs all "slave nodes". In the preferred embodiment, the master node is the network controller 62 and the slave nodes are the STTs 16. The master node initiates protocol migration by sending a migration initiation message to the APP 130 within the slave node, to prepare to receive a new PSP 129.

The master node may initiate migration for all slave nodes, or may select specific slave nodes. The selection of slave nodes to be migrated to the new PSP may occur in several different ways. In the preferred embodiment, the master node polls all slave nodes to determine the type and revision of PSP 128 resident within each node. Each slave node provides the requested information in response to the master node. Alternatively, the master node may execute a routine to interrogate each slave node to determine the type and revision of PSP 128 resident within each slave node. Depending upon the response, or lack thereof, from each slave node, the master node can determine which revision of PSP 128 is resident within each slave node. For example, if a slave node has an old PSP 128 revision, the slave node may be unable to respond to the query posed by the master node. Accordingly, the master node will determine that the PSP 128 resident within that slave node should be updated.

Once the protocol migration process is initiated, the master node sends PSP image frames to the APP 130 within the link level message 217 portion of the link level frames 206 as shown in Figure 9. As shown in step 230 of Figure 10, the APP 130 constructs, or loads, the new PSP image from the received frames 206 in RAM 122 that is not currently being used as directed by the memory manager 132. The memory manager 132 monitors and allocates usage of the RAM 122 by different processing components.

After construction of the new PSP 129 image is complete, the master node directs the APP 130 to enter a verification phase of the new PSP 129 at step 232. The APP 130 processes special test frames sent to it that use the old protocol syntax to encapsulate information represented in the new protocol syntax. The verification is carried out on the frame portion that is utilizing the new protocol to ensure that the new PSP 129 is operating properly. The verification ensures that the new PSP 129 properly processes the frames. If the new PSP 129 fails, at step 236, the APP 130 sends a frame resend request and other error messages to the master node.

The verification is performed utilizing the test frame structure shown in Figure 11. The control field 214 and the control subtype field 215 identify the structure of the link level message field 217. For the verification phase, the control type will be 4/Test New PSP (as shown in Table 2). Table 3 displays the control subtypes associated with control type 4/Test New PSP.

TABLE 3

CONTROL SUBTYPE	INFORMATION
0	Reserved
1	Image Checksum, image checksum value
2	Field Check, number of fields, field length (bits), field value

There are two types of verifications as shown in Table 3; an image checksum and a field check. If the control type is 1/Image Checksum, the APP 130 performs a checksum of the downloaded PSP image to ensure that it is equal to the indicated image checksum value.

If the control subtype is 2/Field Check, the APP 130 performs a field check. As shown in Table 3, the control subtype 215 indicates the number of fields, the length of each field and the value utilized to test the field.

In order to test the fields, the values of each field are sent to the APP 130 along with the new protocol segment which is being tested. The APP 130 parses the new message and performs a calculation for comparison with the values indicated in the message. If they are not equal, an error message will be sent to the master node indicating that the slave node has failed the verification.

Figure 12 illustrates the procedure for testing the new PSP 129 in accordance with control type 4. At step 250, it is determined if an image checksum will be performed. If so, the image checksum is calculated at step 251, and at step 253 is compared to the image checksum value that has been transmitted. Depending upon the results of the comparison, the PSP 129 will pass the test (step 256) or fail the test (step 255). Returning to step 250, if an image checksum is not to be performed, step 252 is entered to determine if a field check is to be performed. If not, the PSP 129 has failed the test (step 255). If a field check is to be performed, step 254 is entered and each field is checked to determine if the value of the field equals the values that have been transmitted. If any of the values are not equal, the PSP 129 has failed the test (step 255). If the values are equal, the PSP has passed the field check (step 256).

Figure 13 is an example of values that could be transmitted in the test frame format in a field check. The frame begins with a sync number 208, a revision number 210, a destination address 212, a source address 213, and a control number 214. Since the control number is 4, the control type will be Test New PSP. The particular type of test is determined by the control subtype 215 which is 2, indicating a field check. The length field 216 indicates six fields to be checked. Each field indicates the length of the field and the value. Within the payload field 224 is included the new protocol segment which has formatted values as if it were a new message. The following fields, included within the link level message 217 as shown in Figure 9, will be checked: a four bit generic flow control (GFC) field 218, an eight bit virtual path identifier (VPI) field 219, a sixteen virtual channel identifier (VCI) field 220, a three bit payload type (PT) field 221, a one bit cell loss priority (CLP) field 222, an eight bit header error check (HEC) field 223, and a variable length payload field 224.

The GFC field 218 is used to ensure fair and efficient access between multiple devices sharing a medium. The VPI field 219 allows a group of virtual connections, called virtual paths, to be identified. The VCI field 220 identifies the individual virtual connections within each virtual path. The PT field 221 is used to distinguish user information and network information. The CLP field 222 permits two priorities of cell to be defined where the network may discard low priority cells under congestion conditions. The value of the HEC field 223 is a function of the header (H). The HEC field 223 provides an eight bit cyclic redundancy check on the contents of the cell header. A detailed description of the function of these fields is outside the scope of this disclosure. (The definition of these fields is consistent with CCITT Recommendations 1.150 and 1.361 (Geneva, June 1992)). This is an example of the fields which are part of a new protocol which is encapsulated within the old protocol format.

If the PSP 129 has failed the verification, the master node will note that an error has occurred and will restart the migration procedure. After a predetermined number of failed migration attempts, the master node will abort the migration procedure and tag the particular slave node for servicing. Thereafter, all tagged slave nodes will be repaired or replaced. If the new PSP 129 passes the verification, the slave node is placed in a standby mode, as shown in Figure 10 at step 234. In this mode, the slave node is able to accept and process incoming frames with the new PSP 129 but is able to default to the old PSP 128. The master node repeats the process with other slave nodes until all slave nodes of interest have gone through migration to the new PSP 129.

Once the new PSP 129 has been installed on all of the slave nodes of interest, the transmission network 10 equipment upgrades associated with the new protocol, if any, may be performed. The master node then sends an activate frame using the new protocol format to all STT 16. In step 238, the slave nodes erase the old PSP 128 and execute the new PSP 129 thereafter. This step may be delayed for a defined period of time after which no node failures occur due to the PSP 129 upgrade. The slave node then exits the PSP upgrade procedure.

Although reference has been made to several different frame formats as examples of the operation of the adoptive protocol, the preferred format is an MPEG2 compliant frame/packet format as shown in Figures 18 and 19. Referring to Figure 18, the DLL packet sublayer frame 600 contains an eight bit sync field 601, a one bit transport error indicator field 602, a one bit payload unit start indicator field 604, a one bit transport priority field 606, a thirteen bit packet identifier field 608, a two bit transport scrambling control field 610, a two bit adaptation control field 612, a four bit continuity counter field 614, and an adaptation field 616 and payload field 618 comprising 188 bytes.

The transport error indicator field 602 is a one bit flag that indicates that at least one uncorrectable bit error exists in the associated transport packet. The payload unit start indicator field 604 has a nominative meaning for transport packets that carry packetized elementary stream (PES) or program specific information (PSI) data as well as private data. A logical one indicates that the payload of this transport packet will commence with a pointer to the start of a message. The transport priority field 606 is a flag to indicate that the associated packet is of greater priority than packets

within the same PID stream that do not have the flag. The packet identifier field 608 indicates the type of data stored in the packet payload. The transport scrambling control field 610 indicates whether or not the payload is scrambled (encrypted). The adaptation field control field 612 indicates whether the transport packet header is followed by an adaptation field and/or a payload. The continuity counter field 614 is an incremental counter which increments with each transport packet with the same PID. The preferred embodiment of the present invention does not use the adaptation field 616 for nonaudio or video elementary streams.

The message sublayer frame 619 of Figure 19 is segmented and transported within the payload field 618 of DLL packet sublayer frame 600. The frame includes an eight bit message type field 620, a one bit reserved field 622, a three bit address type field 624, a twelve bit message length field 626, a forty, sixteen, or twenty-four bit address field 628, an eight bit protocol version field 630, an n -bit application (user layer) data field 632, and a thirty-two bit CRC (cyclic redundancy code) field 634.

The message type field 620 is an eight bit field that distinguishes the type of application and the associated application layer message structure. The MPEG2 table format field 622 is a one bit flag that indicates if the message structure conforms to those previously defined. The address type field 624, is defined as:

- 0 = Broadcast
- 1 = Singlecast type 1:40 bit physical address
- 2 = Singlecast type 2:40 bit logical address
- 3 = Multicast type 1:40 bit address
- 4 = Multicast type 2:16 bit address
- 5 = Multicast type 3:24 bit address

The protocol version field 630 indicates the version of the application message syntax. If the protocol-version number is recognized, the message/frame may be processed, otherwise it is discarded. The message body field 632 may consist of up to 1024 bytes of application layer protocol. The CRC field 634 provides a means of error detection.

During the migration of the STTs 16 to the new PSP 129, a slave node may "crash" or lose communication with the master node. Accordingly, a recovery procedure is provided for reestablishing communication with the master node. The recovery procedure is facilitated by two features: 1) a universal subframe, and 2) a primitive PSP. The master node uses a universal subframe whenever the absence of a previously resident slave node is detected. The absence of a slave node is detected when the master node receives no response to instructions sent to a slave node. The universal subframe comprises a predefined bit pattern recognizable to the slave node when the slave node is attempting recovery. A primitive PSP that recognizes the universal subframe is included within the APP 130.

The universal subframe and the primitive PSP enable downloading of the current PSP, in accordance with the procedure previously described, even when the STT 16 is operating in a degraded condition (i.e. is operating with no other PSP than the primitive PSP). The new PSP image data is appended to the universal subframe to enable the primitive PSP to recognize the frame and deliver the image information to the APP 130 for the construction of the new PSP. After recovery, the slave node has the most current PSP being used by the STTs 16.

The recovery procedure is shown in Figure 14. As the slave node enters the recovery mode, it detects the universal subframe at step 240, and sends the master node a verification message that the subframe has been detected. The APP 130 utilizes the data appended on to the universal subframe to build the new PSP image in the RAM 122 at step 242. After construction of the new PSP 129 is complete, the slave node executes the new PSP at step 244 and exits the recovery mode.

The operation of the adaptive protocol scheme may be best described by using a state transition table representation as shown below. The scheme comprises the following four states: 1) current state, 2) input event, 3) next state, and 4) output event. The current state is a state that the APP 130 is in prior to receiving an input event. An input event is a message from the master node which causes the APP 130 to perform certain actions and proceed to the next state. The next state corresponds to actions taken by the slave node and the following state transition. The output event corresponds to the message generated from the slave node to the master node.

TABLE 4

Current State	Input Event	Next Action/State	Output Event
Use-Current-PSP	Expected event Rx-Update-PSP	Prepare for New-PSP	
	Unexpected events Rx-New-PSP-Frame Rx-Last-PSP-Frame Rx-Test-New-PSP-Frame Rx-Standby-Frame Rx-Activate-New-PSP Rx-Set-New-To-Current Rx-Use-Curr-PSP	Use-Current-PSP	Tx-Event-Out-of-Sync (event, curr-state)
Prepare-for-New-PSP	Expected event Rx-Update-PSP	Prepare-for-New-PSP	
	Expected event Rx-New-PSP-Frame	Build-New-PSP-Image	
	Unexpected events Rx-Update-PSP Rx-Last-PSP-Frame Rx-Test-New-PSP-Frame Rx-Standby-Frame	Act 01: Abort & Cleanup, Use-Current-PSP	Tx-Event-Out-of-Sync (event, curr-state)
	Rx-Activate-New-PSP Rx-Set-New-To-Current Rx-Use-Curr-PSP		
Build-New-PSP-Image	Expected event Rx-New-PSP-Frame	Build-New-PSP-Image	
	Expected event Rx-Last-PSP-Frame	Test-New-PSP	
	Unexpected event Rx-Update-PSP Rx-Test-New-PSP-Frame Rx-Standby-Frame Rx-Activate-New-PSP Rx-Set-New-To-Current	Act 01: Abort & Cleanup Build-New-PSP-Image	Tx-Event-Out-of-Sync (event, curr-state)
	Unexpected event Detect-out-of-Sync-Frame*	Build-New-PSP-Image	Tx-Frame-Out-of-Sync (frame-sequence-no)

Current State	Input Event	Next Action/State	Output Event
	Unexpected event Rx-Use-Curr-PSP	Act 01: Abort & Cleanup, Use-Current-PSP	Tx-Event-Out-of-Sync(event, curr-state)
Test-New-PSP	Expected event Rx-Test-New-PSP-Frame	Test-New-PSP	If test result < > unsuccessful, TX-test-Failure (failure code)
	Expected event Rx-Standby-Frame	Standby	
	Expected event Rx-Test-New-PSP-Frame Rx-Last-PSP-Frame Rx-Update-PSP Rx-New-PSP-Frame Rx-Set-New-To-Current	Test-New-PSP	Tx-Event-Out-of-Sync(event, curr-state)
	Unexpected event Rx-Use-Curr-PSP	Act 01: Abort & Cleanup, Use-Current-PSP	Tx-Event-Out-of-Sync(event, curr-state)
Standby	Expected event Rx-Standby-Frame	Standby	
	Expected event Rx-Activate-New-PSP	Use-New-PSP	
Standby (cont.)	Expected event Rx-Test-New-PSP-Frame Rx-Last-PSP-Frame Rx-Update-PSP-Frame Rx-New-PSP-Frame Rx-Set-New-To-Current	Standby	Tx-Event-Out-of-Sync(event, curr-state)
	Unexpected event Rx-Use-Curr-PSP	Act 01: Abort & Cleanup, Use-Current-PSP	Tx-Event-Out-of-Sync(event, curr-state)
Use-New-PSP	Expected event Rx-Set-New-To-Current	Act 02: Final-Cleanup Use-Current-PSP	
	Expected event Rx-Update-PSP Rx-New-PSP-Frame Rx-Last-PSP-Frame Rx-Test-New-PSP Rx-Standby-Frame Rx-Activate-New-PSP	Use-New-PSP	Tx-Event-Out-of-Sync(event, curr-state)
	Unexpected event Rx-Use-Curr-PSP	Act 01: Abort & Cleanup Use-Current-PSP	Tx-Event-Out-of-Sync(event, curr-state)

Each transition to or from the states in Table 4 has a set of actions associated with it. A transition to a more advanced state is conditional upon getting an expected outcome from the response of each current state to the input event. If an expected state is not achieved, the APP 130 will remain in the current state or revert to a previous state.

The adaptive protocol scheme described herein not only permits upgrading of STTs, but also adaptation of STTs 16 belonging to one network to another new network, where the latter uses a different communication network infrastructure. This eliminates the need for maintaining two or more separate networks and enables all systems to migrate to the latest system, provided the STT 16 hardware does not require any changes. This approach eliminates the backward capability limitations imposed by fixed protocols by enabling the migration of existing systems to new protocols to take advantage of new, more enabling protocols in a cost effective manner.

The preferred embodiment of the network architecture also employs a hybrid medium access control (MAC) system 400 to control access to the upstream bandwidth by the plurality of STTs 16. The hybrid MAC system 400 comprises different MAC components which are selected based upon the type of communication that is associated with the corresponding application or service selected by the VIU. Each MAC component resides on a separate frequency. MAC parameters are configurable to provide additional flexibility and operational and resource optimization (such as latency, probability of blocking and bandwidth utilization).

Applications and services selected by the VIU can be categorized according to the communications required to support the application or service. The first category of applications and services are associated with asynchronous, latency independent communications. These communications are the least demanding from a performance standpoint since the applications and services are capable of functioning effectively with response time latencies of minutes or hours. This is typical of transactions that do not require a subsequent action to be performed by the application or the network 10 in support of a VIU visible response to an original action. For these applications and services, the time that it takes to deliver the data to the associated destination is not a critical factor.

The second category of applications and services are associated with asynchronous, contention-prone communications. These communications place medium demands on performance and thus, maximum response latencies on the order of sub-seconds are required. This category includes transactions and services that are followed by a subsequent action at the application, the L1G 20, or the VIP 12 in support of a VIU visible response to an original action. Timely delivery of information on the order of microseconds is critical for applications such as information on demand and video-on-demand applications.

The third category of applications are associated with isosynchronous communications. These are the most demanding in terms of bandwidth, guaranteed maximum latencies (milliseconds), and a symmetric utilization of upstream and downstream bandwidths. Advanced multi-media applications, plain old telephony service (POTS), personal communication services (PCS), and video telephony are some of the applications in this category. By allocating a separate portion of the bandwidth for different types of communications, the hybrid MAC system 400 of the present invention will support future services that are not yet envisioned.

Referring to Figure 16, the configurable hybrid MAC system 400 employs space division multiplexing, frequency division multiplexing, and time division multiplexing to efficiently utilize the upstream bandwidth. The space domain allows for dividing the subscriber population into smaller segments based on a number of factors such as physical location, subscriber population density and projected growth. Each network controller 62 may be assigned a number of physical signal transmission paths, such as a fiber 90 and associated carriers. This allows for gradual network expansion from the physical topology standpoint since additional network controllers 62 may be added to the headend 18 based upon the performance needs of the network which are a function of fiber to the node (FTTN) size, service take rate, or simultaneous usage of a given service, etc.

The frequency domain facilitates the efficient support of the hybrid MAC components as well as further segmentation of the settop terminal 16 population. Each of the hybrid MAC components are allocated a portion of the upstream bandwidth having at least one RF carrier frequency. This is a primary factor behind the simplicity of the design that could not be achieved by prior art MAC methods that attempted to support asynchronous, anisochronous and isosynchronous data without distinct channel separation.

The time domain is used to allow multiple access at the smallest granularity as compared to the other two domains. The time domain is the basis for the fixed and both types of dynamic TDMA MAC components, (dynamic multi-rate TDMA and random slot reservation-dynamic slot assignment (RSR-DSA) TDMA). Multiple access within the same STT 16 population segment and the same allocated upstream frequency is achieved in this domain.

The hybrid MAC system 400 is adjustable to the variety of current and future applications that may be added from a network resource support standpoint by categorizing each application (as aforementioned) according to the network resources required to support the following types of communications: 1) isochronous communications; and 2) asynchronous communications including, latency independent and contention-prone communications.

In operation, the network controller 62 receives a plurality of service requests over preassigned default connections from the plurality of STTs 16. These requests are forwarded to the network controller 62 through the QPSK demod/mux 60. Once allocation of resources (frequency and time domain) are selected by the network controller 62, confirmation is sent downstream to the MAC module 126 via OBTM through an appropriate QPSK mux/mod 58. The process is repeated by other STTs 16 as application and service session requests are made.

Referring to Figure 5, the preferred STT 16 for implementing the hybrid MAC system 400 is shown. The STT 16 includes a medium access control module 126 and a decryption/encryption control module 124 for communicating with

the network controller 62 through the transmission network 56. The STT 16 initiates a service request when the VIU selects a particular application, service, or session. The MAC module 126 analyzes, based on well known application service types, the communication requirements (i.e. isosynchronous, asynchronous) of the requested service or session type, and selects the MAC component which will most closely meet the communication requirements. Since each

Alternatively, a network controller client, such as ASEM 22, may request (on behalf the L1G) the allocation of resources in support a session of a given service. Once the network controller 62, allocates the appropriate MAC resources, it informs the STT 16 of the relevant MAC parameters associated with the connection established in support of the requested session.

Referring to Figure 15, a TDMA frame 410 as used by the dynamic multi-rate TDMA MAC component is shown. The frame 410 comprises a series of time slots 412 which may be separated by in time 414. The configurable parameters for the dynamic multi-rate MAC component are the frame size, the time slot size and the spacing 414 between time slots 412. The size of the frame 410 may be varied by increasing or decreasing the number of time slots 412. This affects the latency and the effective transmission rate of the communication. The size of the time slot 412 may also be varied to change the number of clock cycles within a time slot, and therefore the number of packets, that may be transmitted in a given time slot 412 (again, changing the effective transmission rate). The spacing 414 between time slots 412 may be varied to change the number of packets which may be transmitted within a given frame 410. The adjustment of the spacing 414 between time slots 412 is also used for propagation delays to compensate for the different physical distances of the STTs 16 from the network controller 62.

The static TDMA MAC component is typically assigned by the network controller 62 based on a connect request from a network controller client such as ASEM 22. For the static TDMA MAC component, the TDMA frame is fixed at one time slot per STT 16. This provides guaranteed bandwidth for access by STTs 16 within a given frame time conveying additional connection requests or conveying diagnostic/status information which may also be forwarded on the POP carrier).

In the preferred embodiment, the RSR-DSA MAC component is the default TDMA channel for large nodes with high take rates in place of the static TDMA. If the MAC module 126 has determined that the requested service is suited for random slot reservation - dynamic slot allocation (RSR-DSA), the STT 16 initiates upstream communications by randomly selecting a time slot 412 within a given frame 410 or by transmitting on a previously reserved time slot 412. The probability of collisions is inversely proportional to the number of time slots 412 per frame 410 and directly proportional to the number of STTs 16 that need to communicate during that cycle. The network controller 62 assigns the RSR-DSA TDMA frames 410 in accordance with the procedure shown in Figure 17A. In step 300 the STT 16 may reserve a time slot 412 once, or may request reserving the time slot 412 over multiple cycles. This reservation request is transmitted upstream over a randomly selected time slot 412 (step 302). If a collision is detected by the network controller 62, the network controller 62 transmits a message to the STTs 16 to retransmit the request (step 303). If no collision is detected by the network controller 62, the network controller 62 receives the request (step 304) and sends an acknowledgment to the requesting settop terminal 16 and to all other STTs 16 that the particular slot within that channel is no longer available (step 306). The STT 16 receives this acknowledgment (step 308) and the requesting STT 16 beings communications. The network controller 62 dynamically assigns the requested time slot 412 by acknowledging a reservation by a message sent to the STT 16 in the downstream OBTM. The same acknowledgment indicates to other STTs 16 that the time slot 412 is no longer available. After the STT 16 has terminated communications (step 310), the time slot 412 is released. The network controller 62 monitors the channel activity (step 312), and when it determines (step 314) that the time slot 412 has been released, a "time slot free" message is sent to all STT 16 (step 316.) All STTs 16 receive an acknowledgment that the time slot 412 is available (step 318). If, while monitoring channel activity (step 412), the network controller 62 determines that there is too much activity over a particular channel (step 320), the network controller 62 adjusts the number of time slots 412, and increases the size of the frame 410, adjusts the spacing 414 between time slots 412 or allocates additional frequencies (step 322). If the increased frame size is greater than the corresponding fixed TDMA frame size, (stem 324) the network controller 62 determines that particular channel is to be designated a static TDMA channel (step 426).

While the RSR - DSA describes the connectionless mode of operation intended for the default TDMA channel in a large node size and high-take rate, the connection-oriented mode is similar in the sense that a time slot 412 or set of time slots may be reserved over multiple cycles (frames) for the duration of the connection. Additional information may be conveyed by the network controller 62 during connection establishment and connection release phases.

In the connection-oriented mode, where multi-rate dynamic TDMA is used, the network controller 62 may assign a frequency and a time slot(s) for the duration of the connection, (typical of applications benefiting from guaranteed bandwidth hence predictable latency/in support of interactive applications). Referring to Figure 17B in this mode, a network controller client (e.g. ASEM 22), or a network module 70 client (e.g. an application within the STT 16) forwards a con-

nection request (step 700) to the network controller 62 specifying which STT 16 is to be connected and the associated TDMA rate desired for that connection. (480 bps, ... 2400 bps ... 32 bps ... 142k, 11520, 16000, 19200, 15360, 32000, 56000, 64000, 112000, 128000, 192000 bps). This rate is a function of the level of interactivity which is characteristic of the application. For isosynchronous services (video, telephony or POTs), the rates depend on the service itself (e.g. video, telephony using 64 kbps or 128 kbps).

The network controller 62 checks the available resources (step 702) to determine if the new request can be accommodated. If required resources (e.g. number of time slots 412) are not available (on any of the frequencies supporting the requested rate or lower multiples of that rate), the network controller 62 rejects the connection request specifying the reason (e.g. no available resources). If on the other hand the network controller 62 determines that the required resources are available (step 704) it reserves the selected time slots 412 within the appropriate frequency (step 708), informs the network module of such parameters, and returns to the client a confirmation to the connection request (step 710).

When the connect request is originated by ASEM 22, the request is forwarded on the network controller's 62 Ethernet port. When the request is forwarded by the STT 16, the network module 70 may forward the request on the defined static TDMA channel or the RSR-DSA default TDMA channel (whichever is employed at the time within a given system).

The plain old polling component of the MAC system 400 is intended for applications that benefit from a store and forward system where collisions are expected but the latency associated with POP response is irrelevant. Additionally, it facilitates controlled communications for any diagnostic operation and presents a fall back communication method for diagnostic purposes should other components of the hybrid MAC system 400 fail.

If the MAC module 126 determines that POP is required, the data transmitter 142 is tuned to the POP frequency and a service message is placed in the polling queue 204.

The polling service may be initiated periodically by the network controller 62 to instruct the STT 16 to purge their polling queue, or a polling request may be transmitted by the STT 16. The polling queue 134 operates as a first-in-first-out (FIFO) buffer which resides in the settop RAM 122. When an application or service requires to send a message upstream, the information is forwarded utilizing resources assigned to the connection supporting the application session. If the information is in response to a poll, the message is copied into the polling queue 134.

The STT 16 transmits the FIFO entries when it receives the polling token, which is an instruction by the network controller 62 for the STT 16 to empty its transmit queue 204. The polling token is broadcast on all OBTM downstream channels and the STT 16 transmits on the polling channel frequency. When the network controller 62 carries out the polling cycle, it has no knowledge of the OBTM downstream frequency to which the STT 16 is tuned. This necessitates sending the polling token on all downstream OBTM carriers for a given neighborhood. Once a STT 16 gets the token, it is permitted to empty its entire polling queue 134. The objective is to minimize the number of individual polling cycles required to empty the STT's entire queue 134. The network controller 62 then reads the messages from the plurality of STTs 16.

The hybrid MAC of the present invention supports of isochronous and anisochronous multi-media communications data in a cost effective and efficient manner. In addition, connection-orientated as well as connectionless services have been supported, where the former requires a guaranteed bandwidth allotment over the duration of the connection.

Although the invention has been described in part by making detailed reference to certain specific embodiments, such details is intended to be instructive rather than restrictive. It will be appreciated by those skilled in the art that many variations may be made in the structure and mode of operation without departing from the spirit and scope of the invention as disclosed in the teachings herein.

Claims

1. A communication system network comprising:

- a remote hub;
- a plurality of head ends associated with said remote hub via a fiber optic interface;
- a plurality of user terminals associated with each head end via a hybrid fiber coax interface;
- said remote hub comprising:
 - means for communicating with interactive and broadcast video information providers (VIPs);
 - means for formatting broadcast signals received from selected VIPs of the broadcast VIPs for direct transmission to the head end/user terminal interfaces such that the user terminals receive said formatted broadcast signals without reformatting by said headends;
 - means for selectively controlling distribution of communication signals between said interactive VIPs and said headends; and
 - means for tracking usage of VIP services by said user terminals for at least billing purposes; and
- each said headend including means for facilitating interactive VIP communications between its associated user terminals and said VIPs.

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2. The system of claim 1, wherein communications over said headend/user terminal interfaces are at least frequency multiplexed into a plurality of downstream channels from said headends and upstream channels to said headends such that the number of downstream channels is substantially greater than the number of upstream channels.
- 5 3. The system of claim 2 wherein said upstream channels are paired with selected downstream channels and communications on at least selected channel pairs are time division multiplexed.

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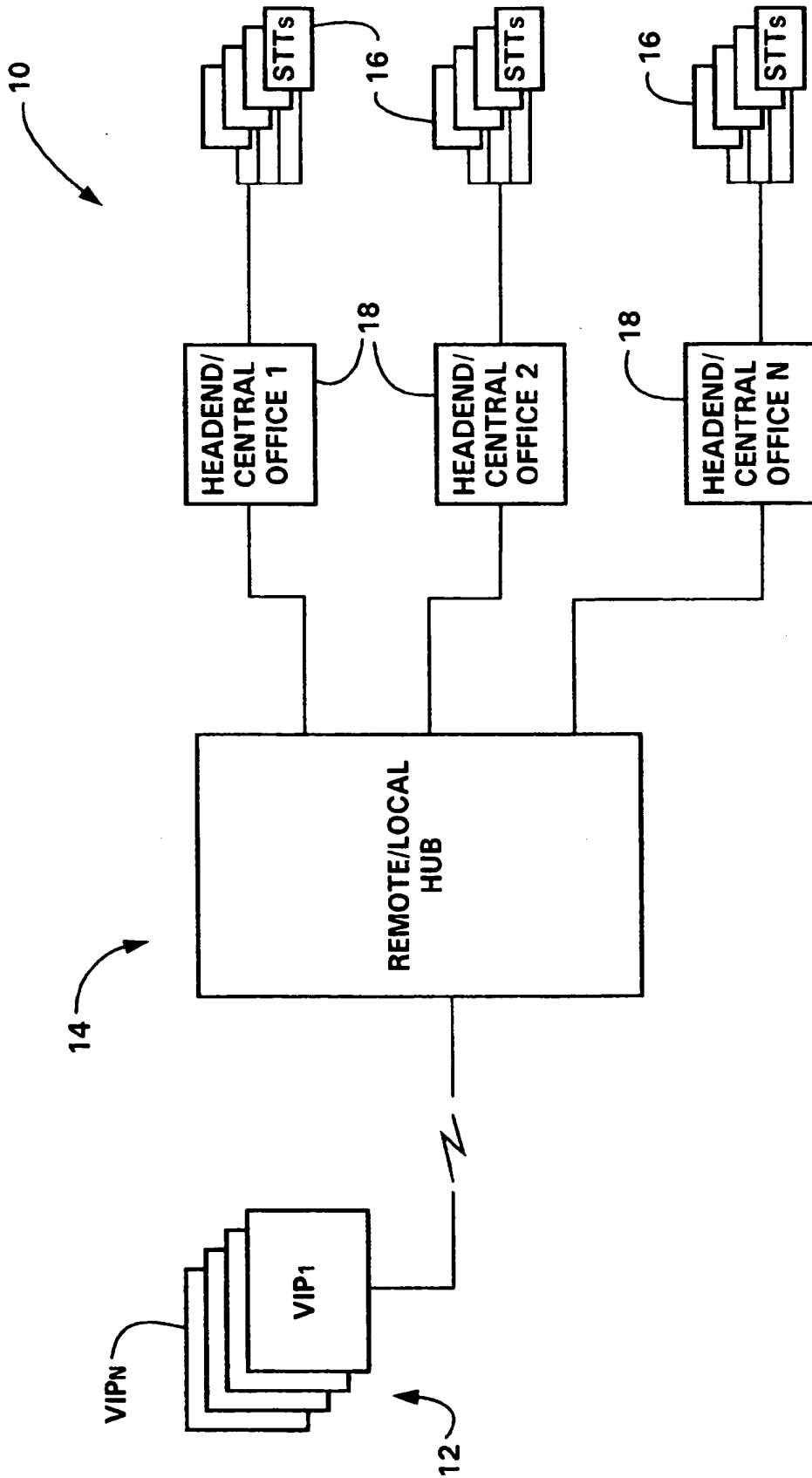
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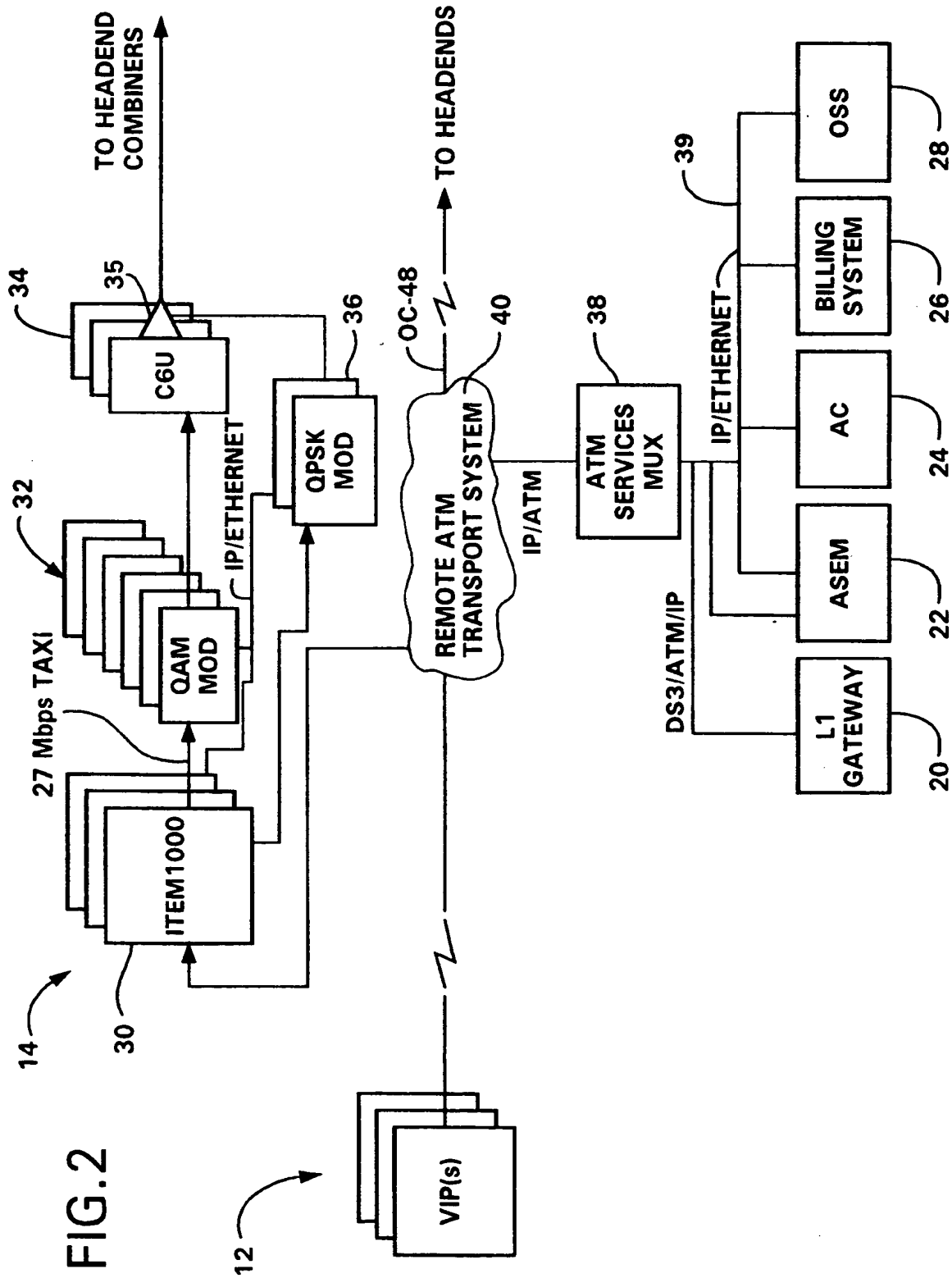
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FIG. 1





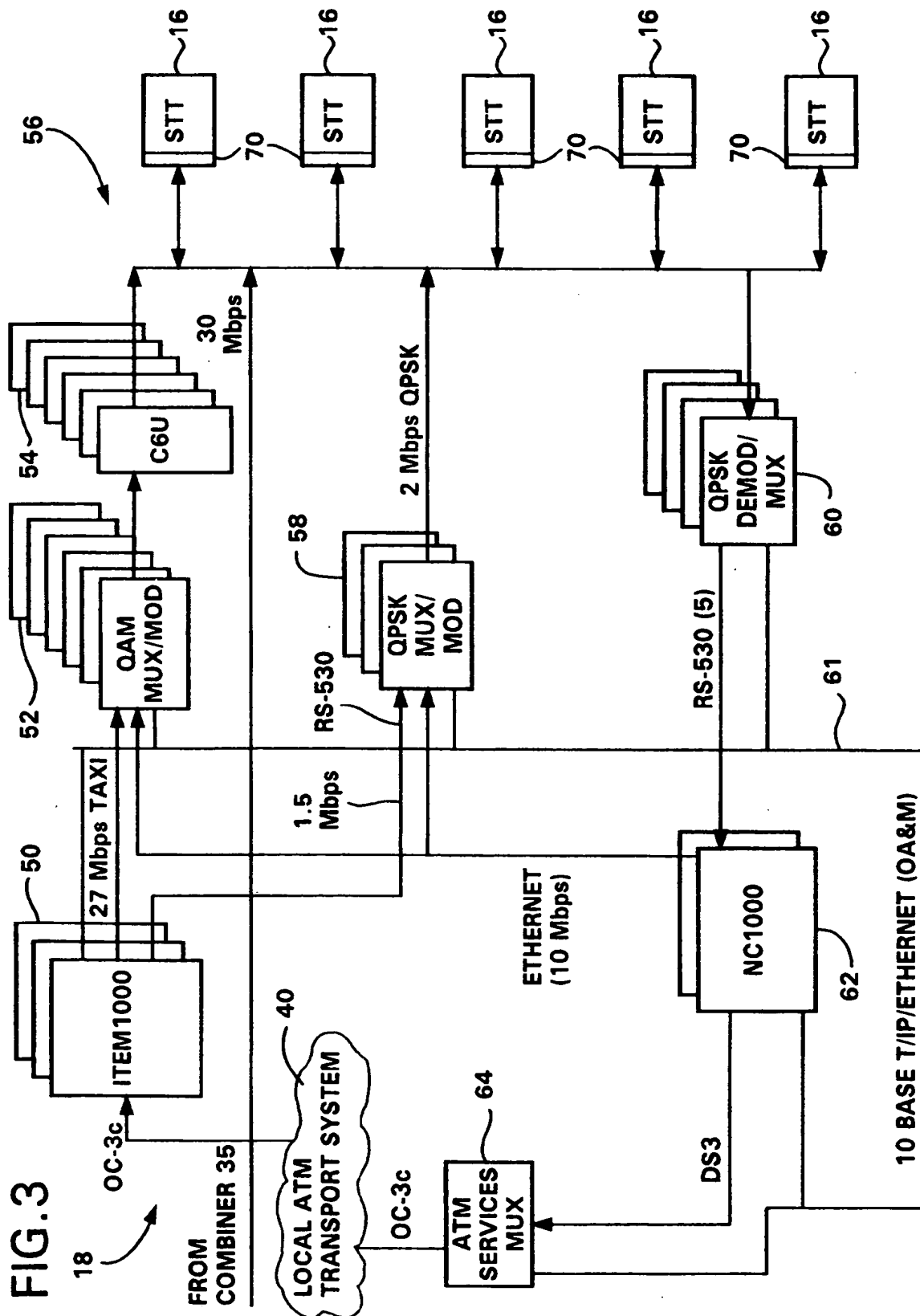


FIG.4

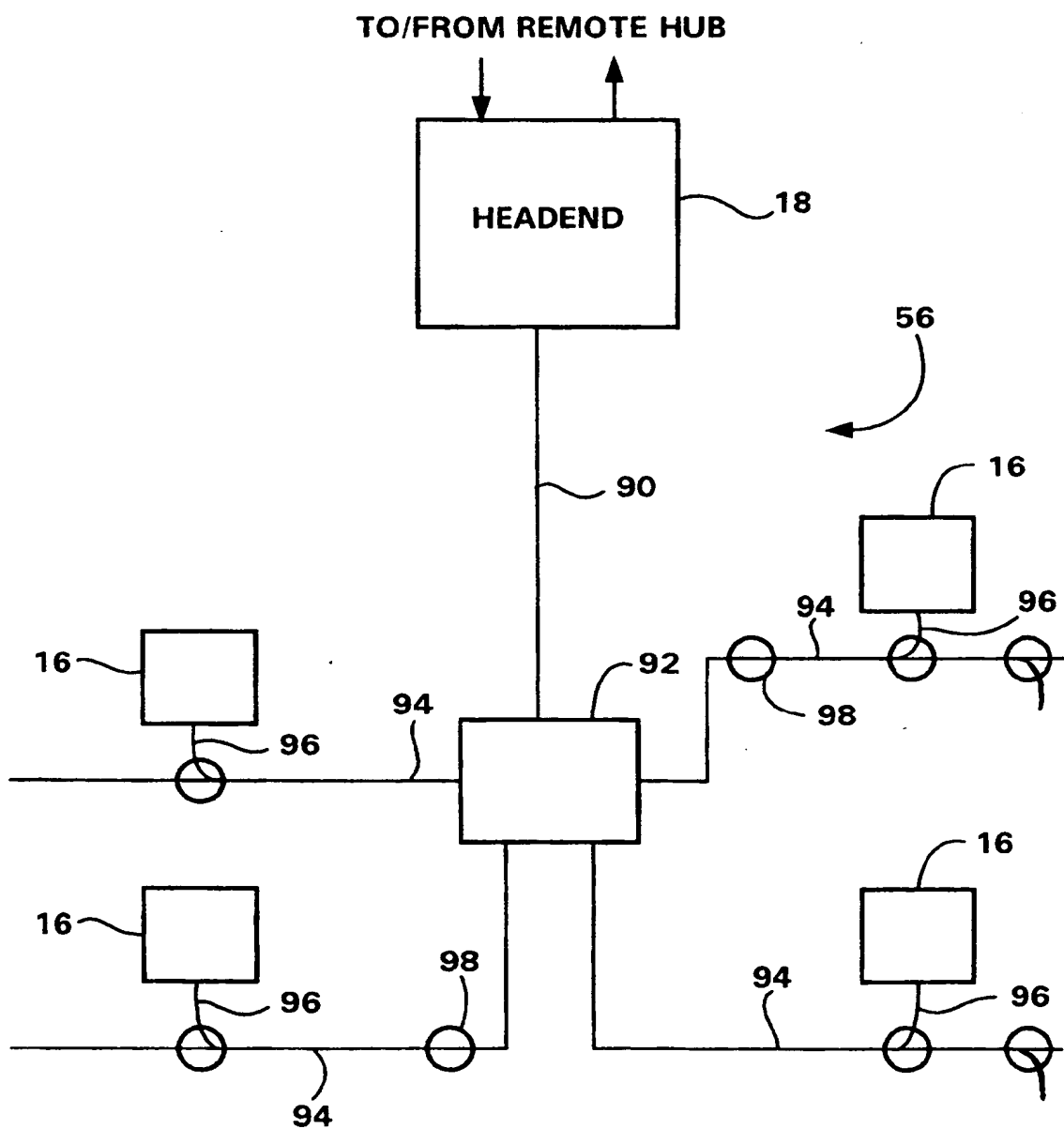


FIG. 5

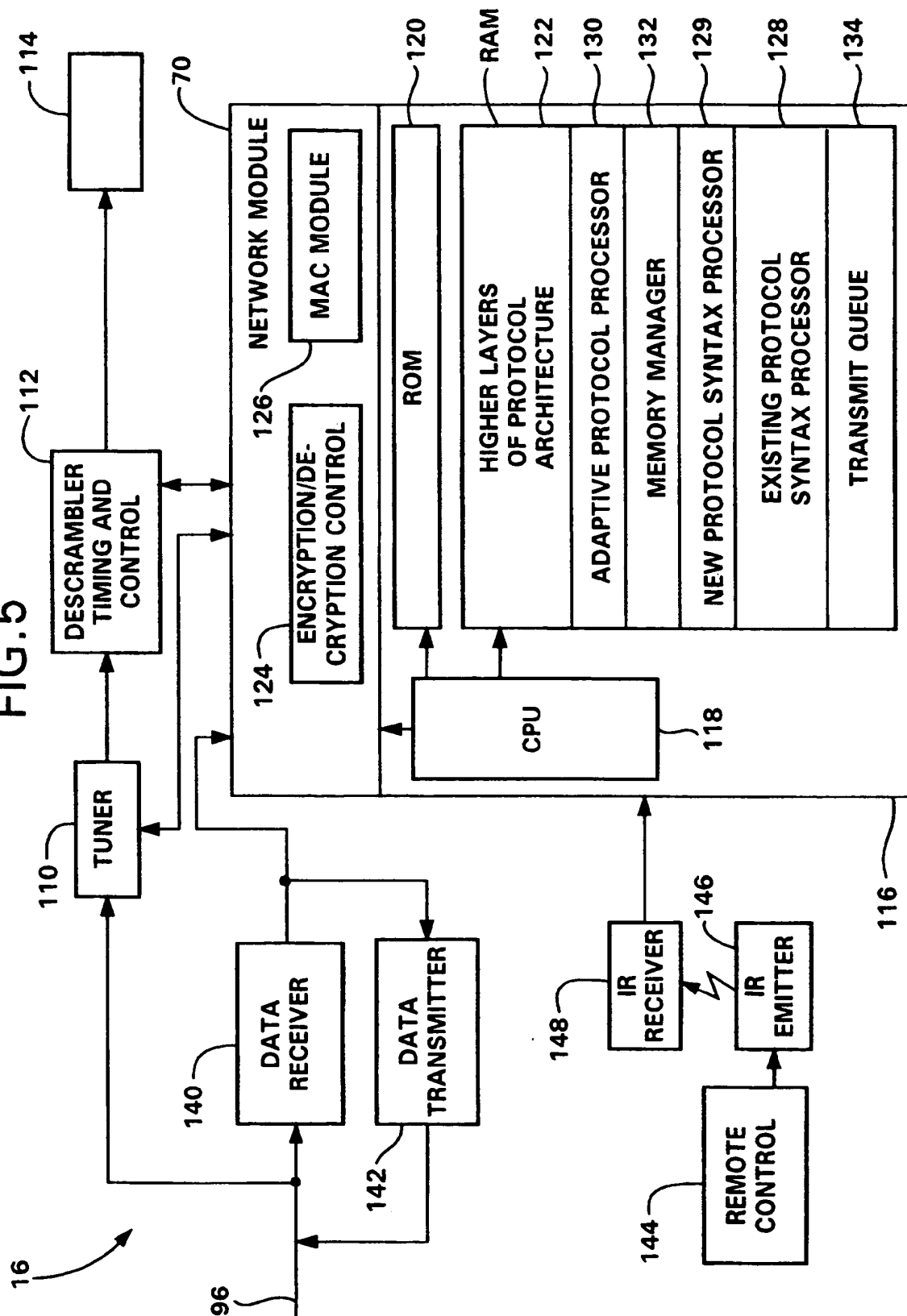


FIG.6
PRIOR ART

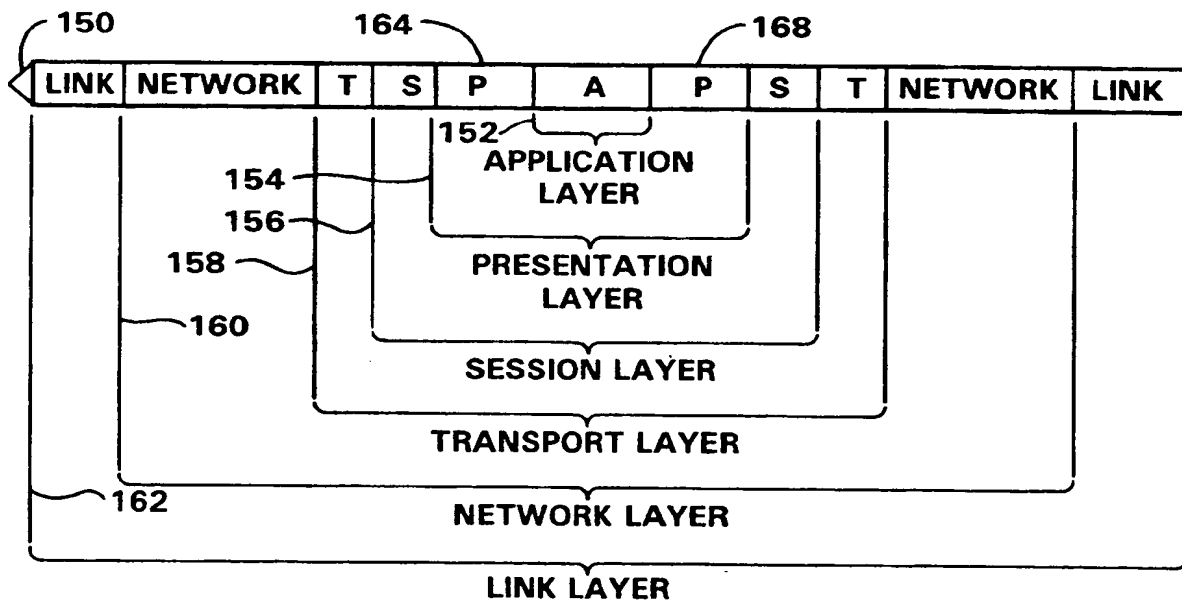


FIG.7
PRIOR ART

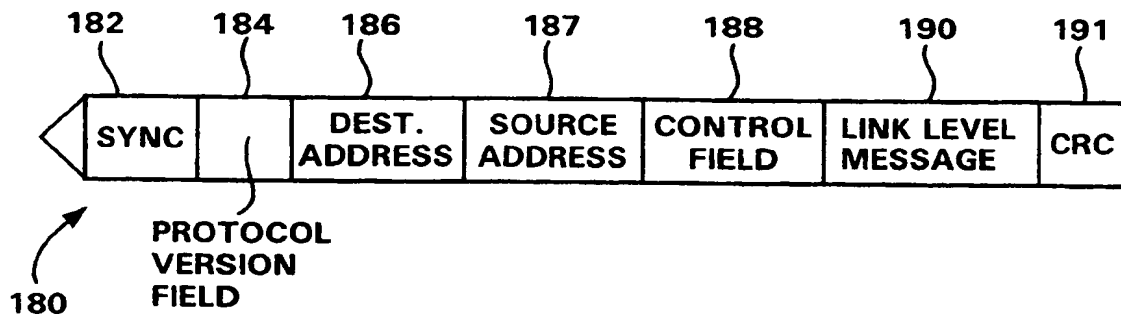


FIG. 8

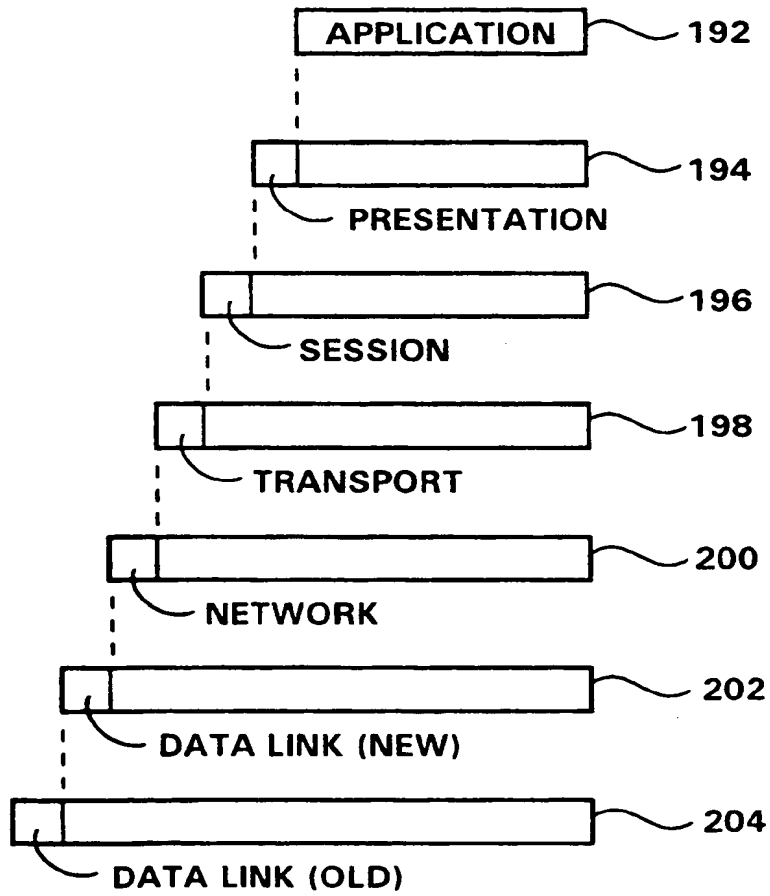


FIG. 9

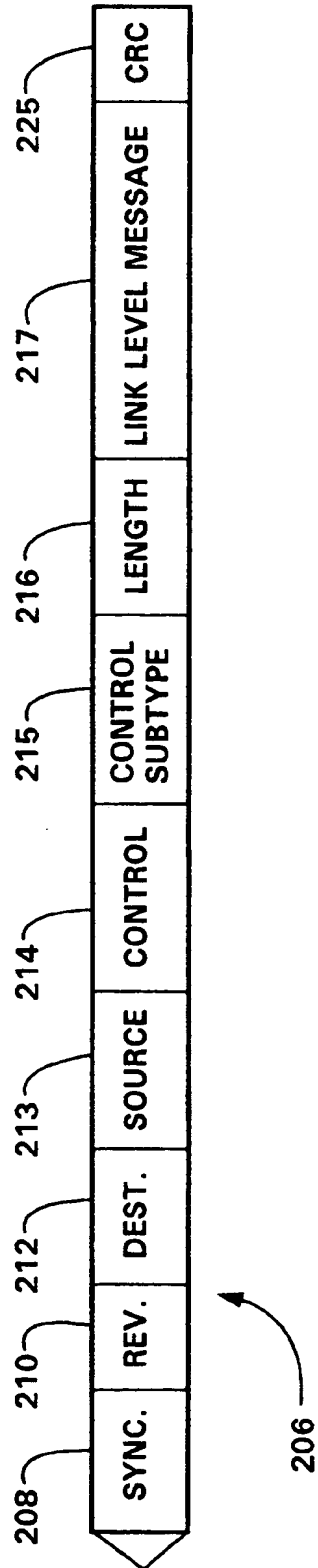


FIG. 11

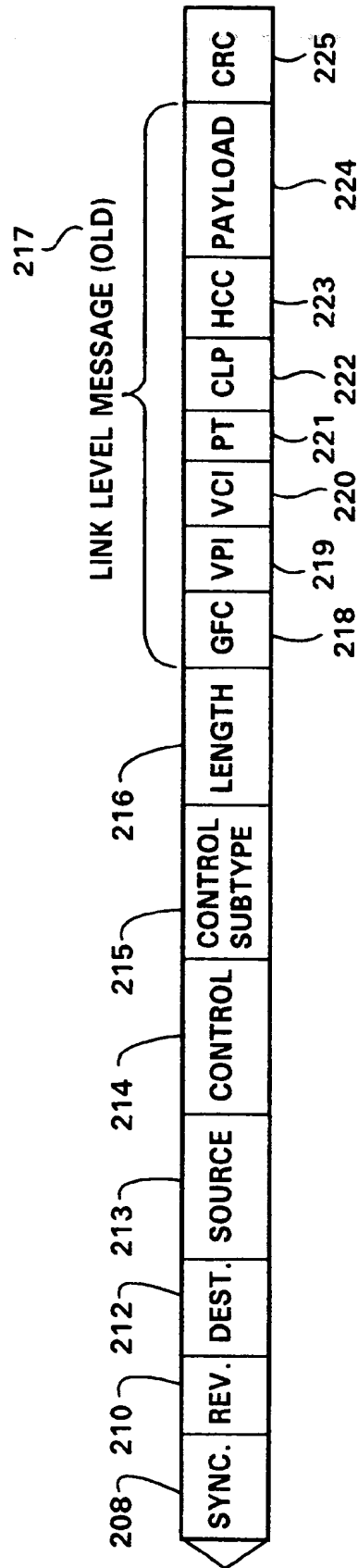


FIG. 10

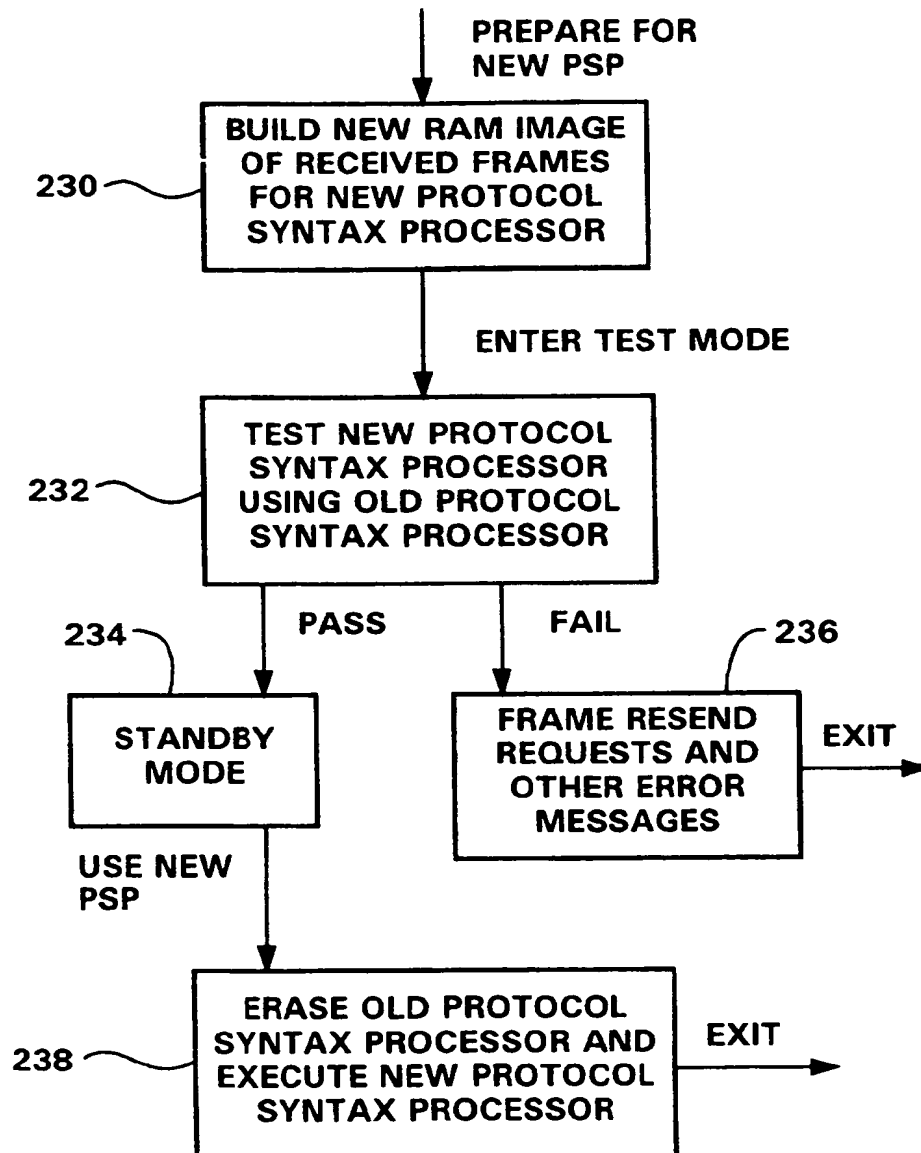


FIG.12

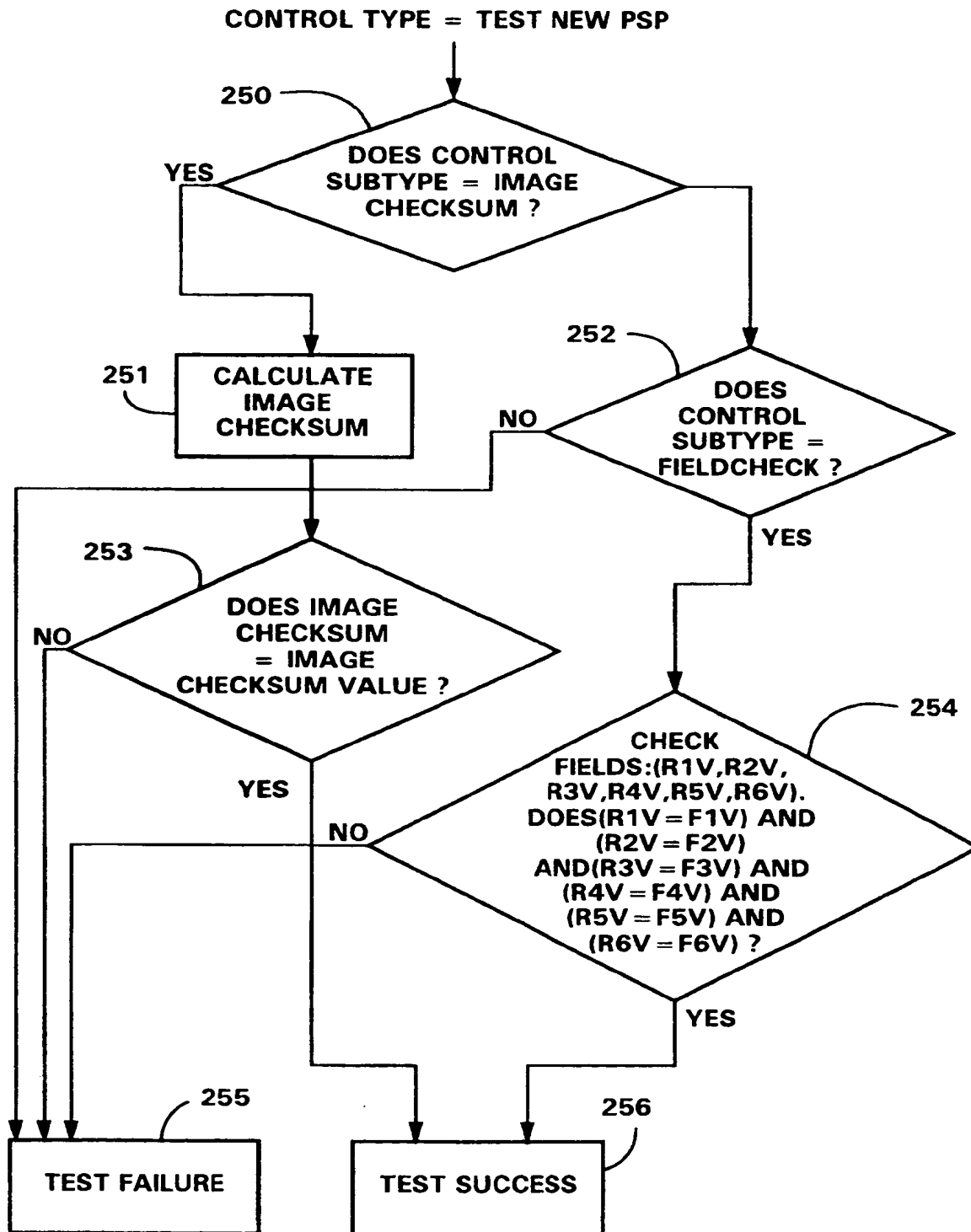


FIG. 13

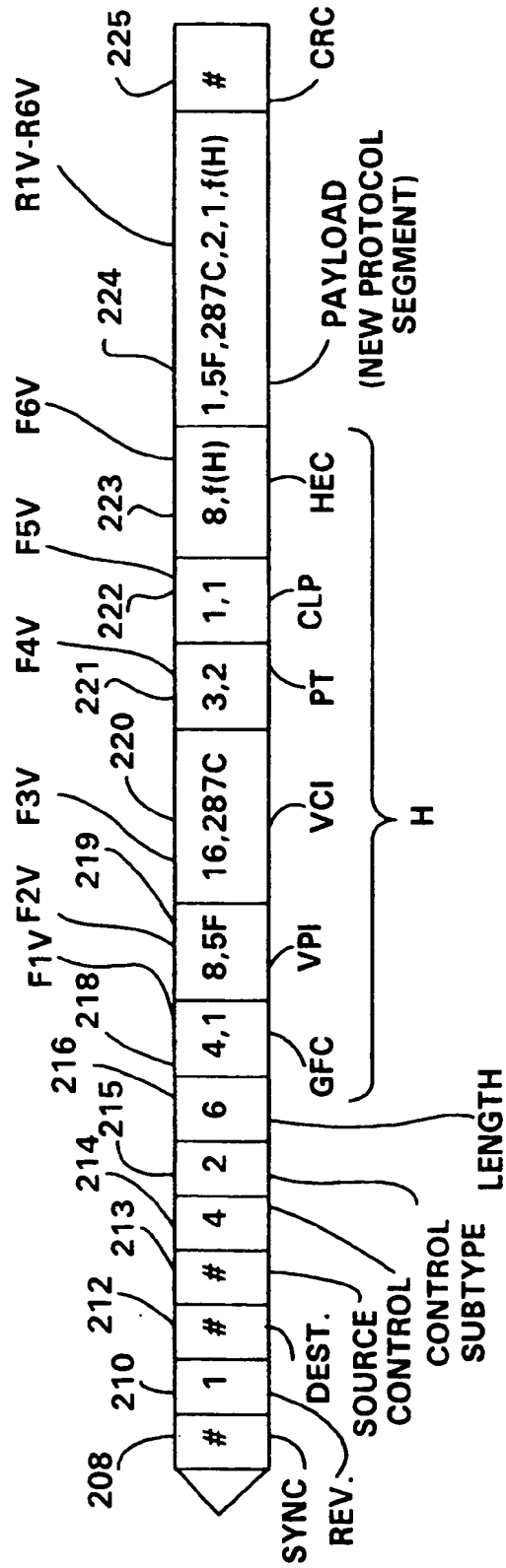


FIG.14

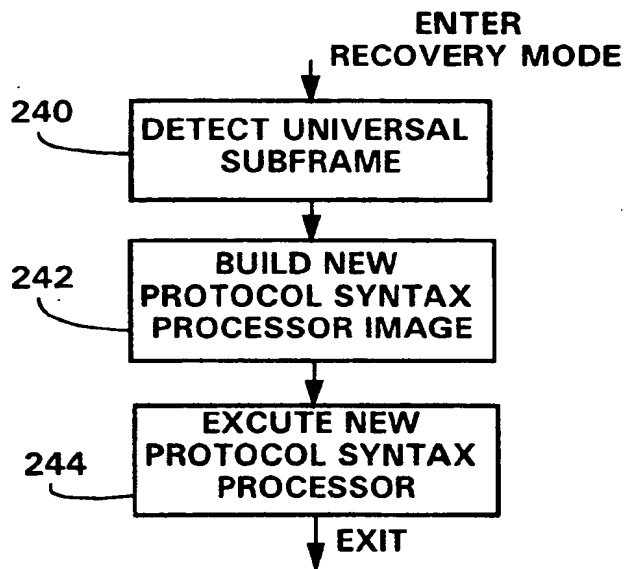


FIG.15

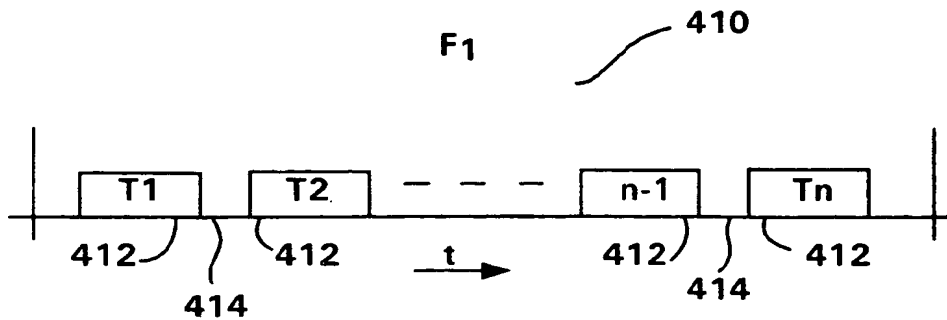


FIG. 16

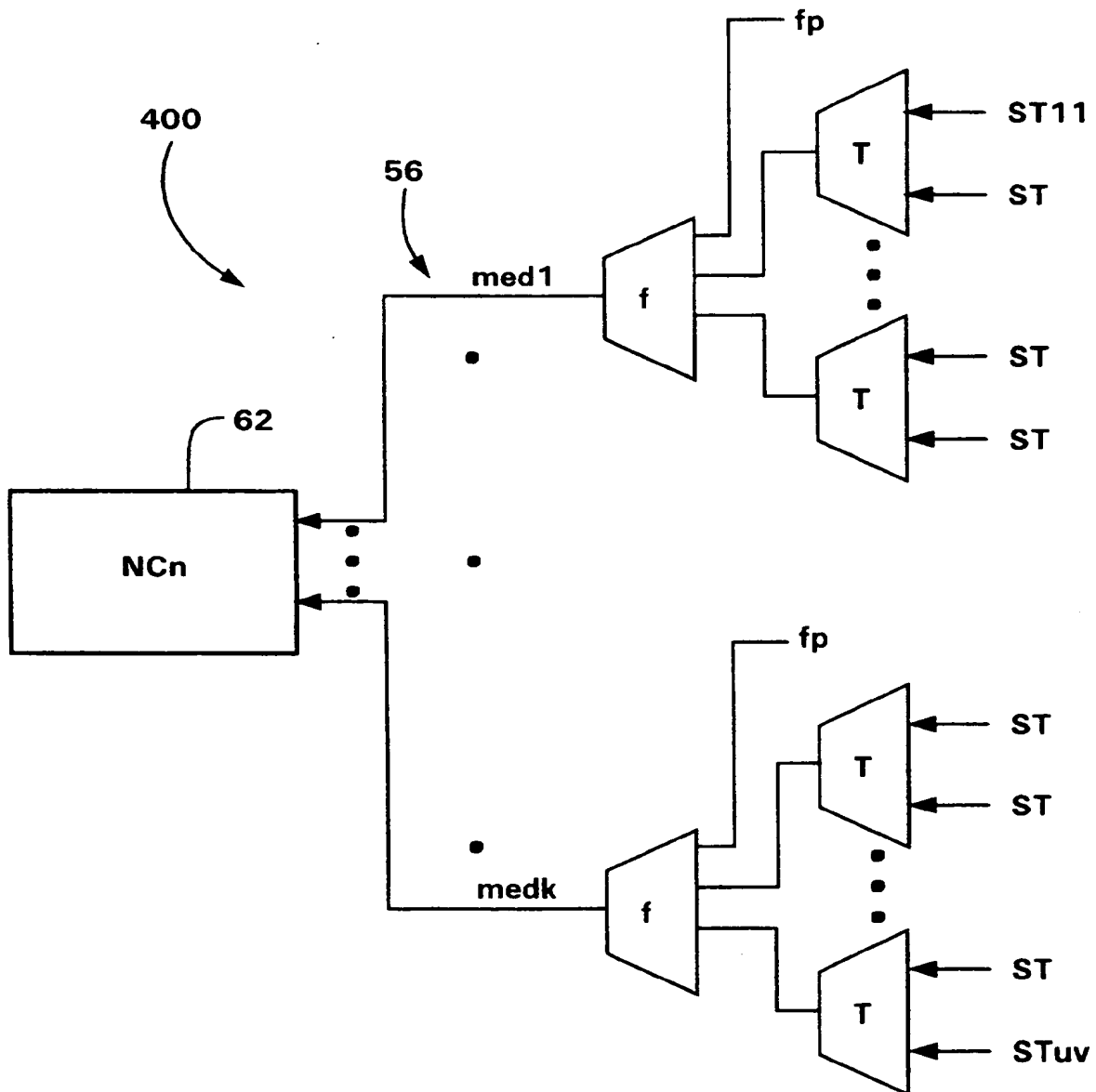


FIG.17A

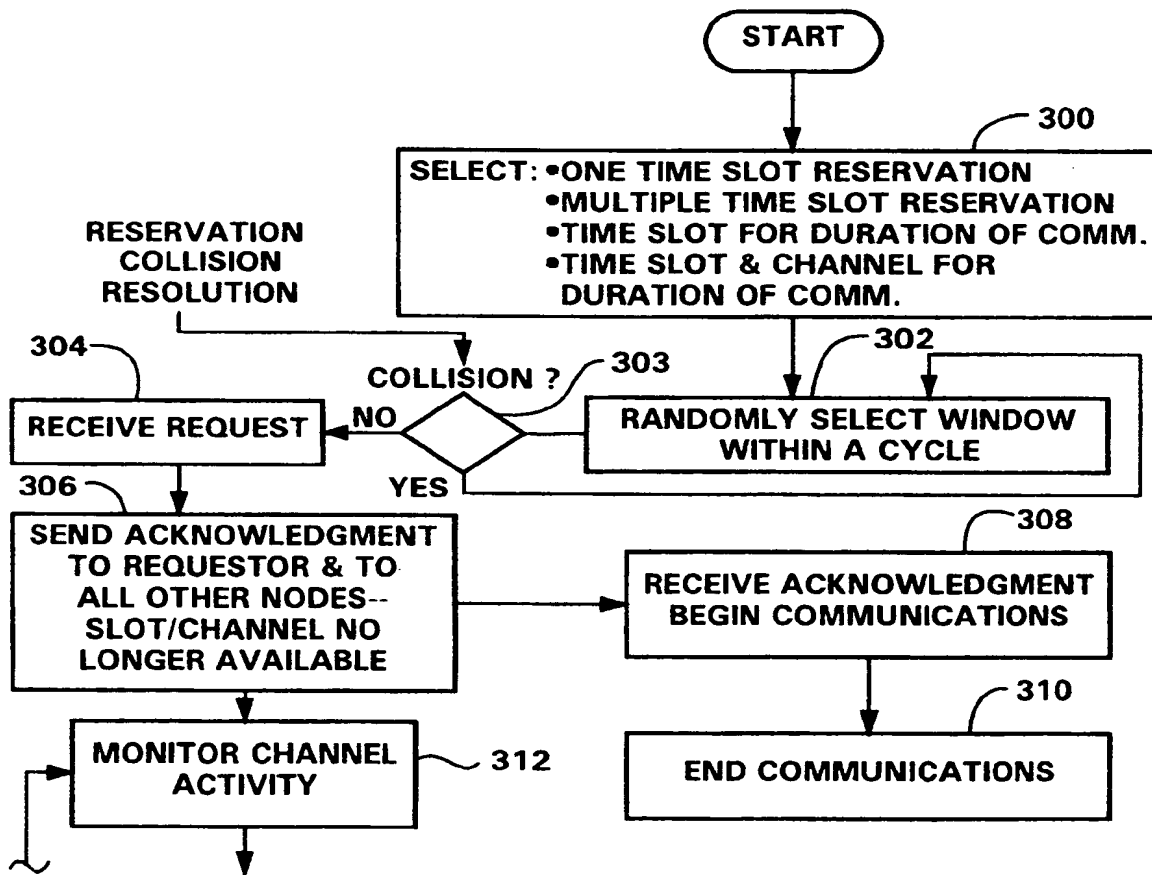
NETWORK CONTROLLERSETTOP TERMINALS

FIG.17A-1

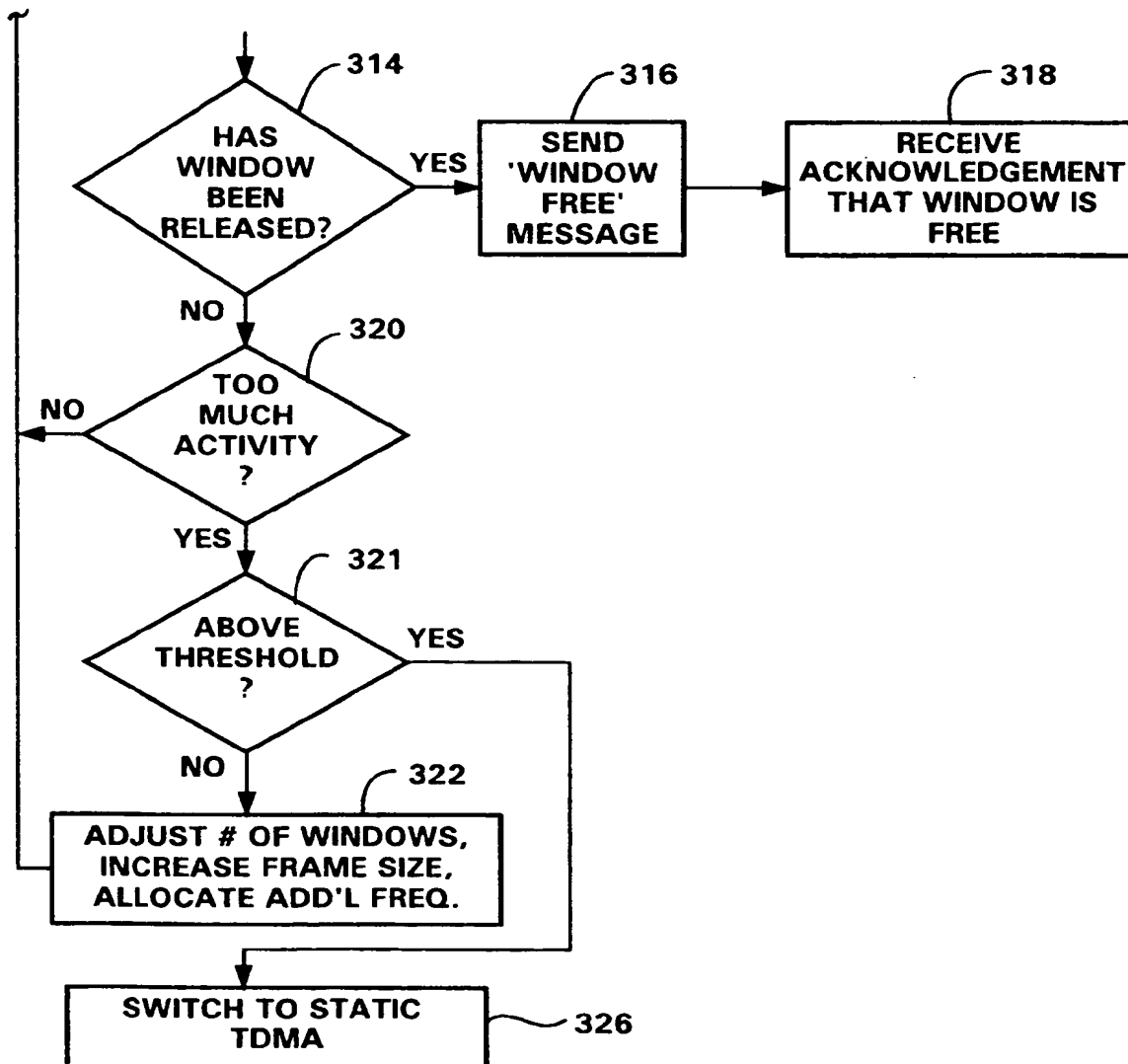
NETWORK CONTROLLERSETTOP TERMINALS

FIG.17B

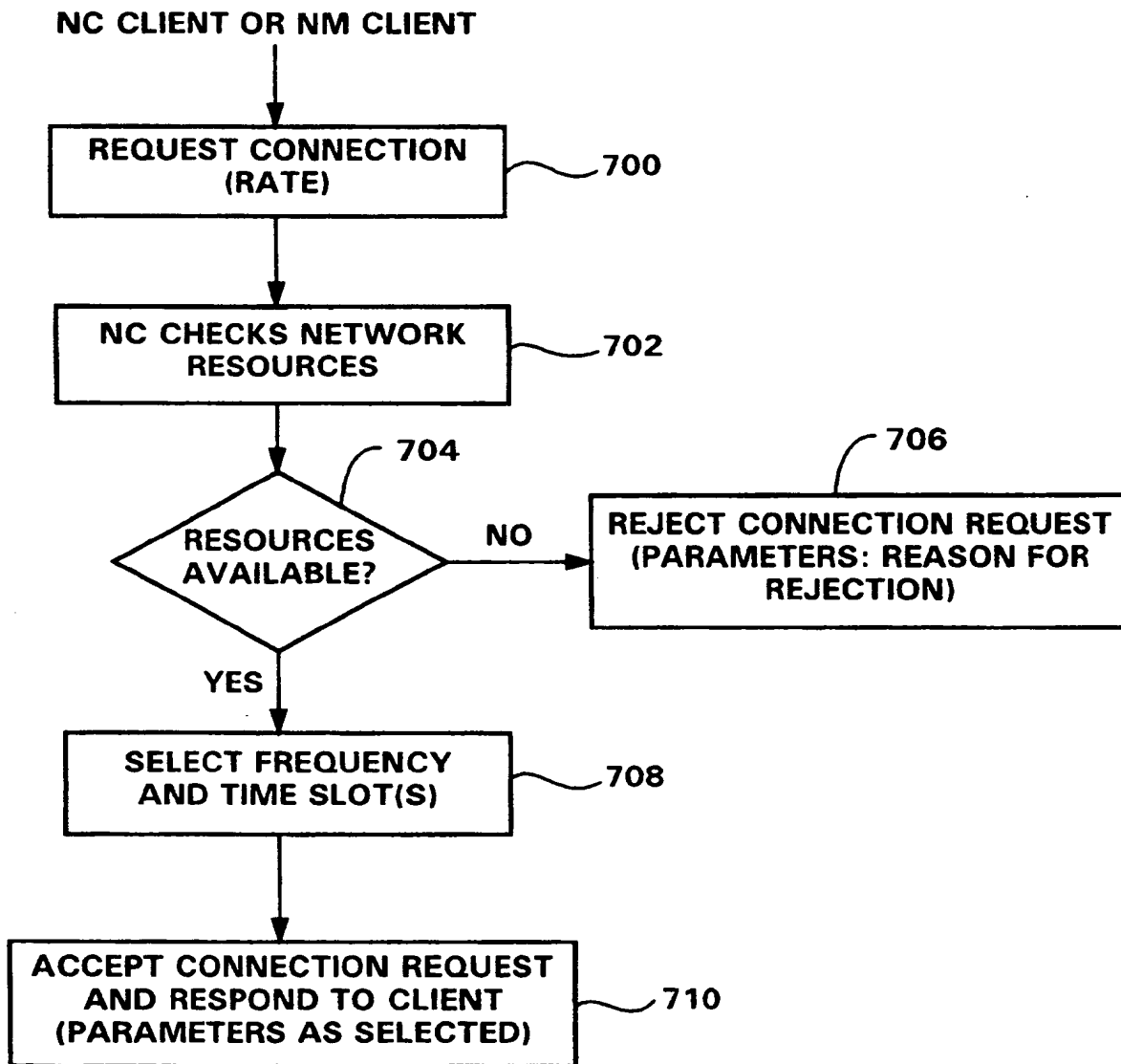


FIG. 18

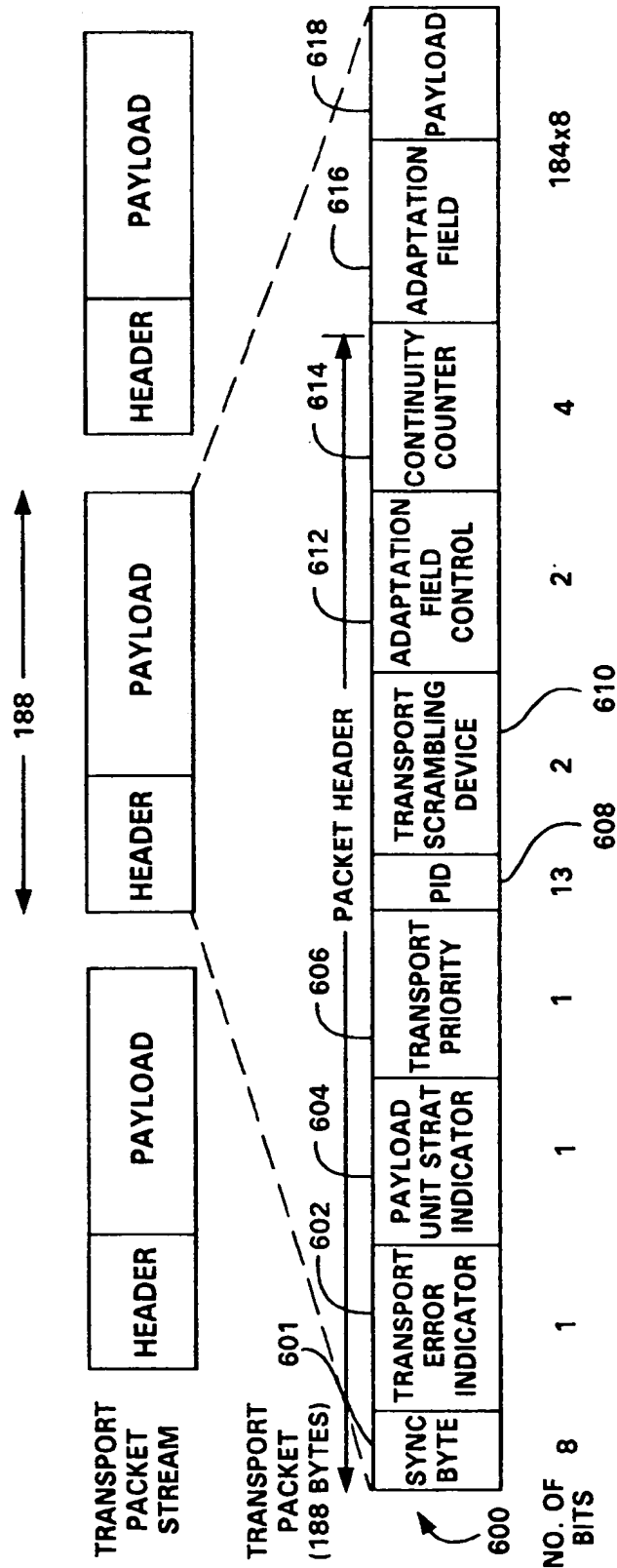
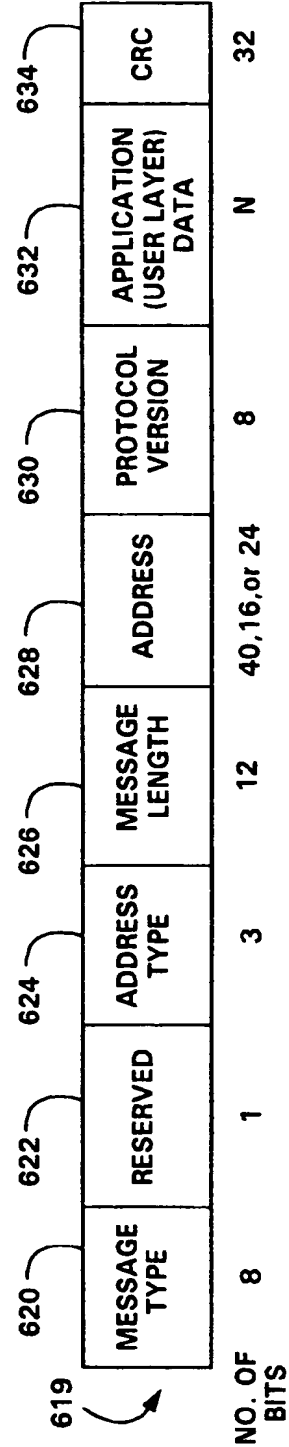
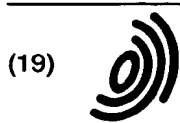


FIG. 19



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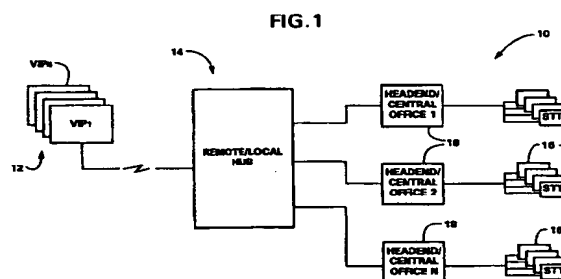
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(54) **Distributed system architecture for digital broadcast and interactive services**

(57) A cable television communication system network (10) comprising a remote hub (14) which is in communication with a plurality of head ends/central office (18) via a fibre optic interface, a plurality of user terminals (16) associated with each head end via a hybrid fibre coax interface and a plurality of interactive and broadcast video information providers (VIPs) (12) which are in communication with the remote hub (14). The remote hub includes means for formatting broadcast signals received from selected VIPs of the broadcast VIPs (12) for direct transmission to the head end/user terminal interfaces such that the user terminals (16) receive said formatted broadcast signals without reformatting by said headends, means for selectively controlling distribution of communication signals between said interactive VIPs and said headends (18), means for tracking usage of VIP services by said user terminals for at least billing purposes. Each said headend (18) means for facilitating interactive VIP communications between its associated user terminals (16) and said VIPs (12).





European Patent
Office

EUROPEAN SEARCH REPORT

Application Number
EP 96 30 1370

DOCUMENTS CONSIDERED TO BE RELEVANT			
Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (Int.Cl.6)
Y	PROCEEDINGS OF THE INTERNATIONAL CONFERENCE ON COMMUNICATIONS (ICC), GENEVA, MAY 23 - 26, 1993, vol. 1 - 2 - 03, 23 May 1993, INSTITUTE OF ELECTRICAL AND ELECTRONICS ENGINEERS, pages 848-852, XP000371202 GELMAN A D ET AL: "AN ARCHITECTURE FOR INTERACTIVE APPLICATIONS"	1	H04N7/173
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Y	IEEE COMMUNICATIONS MAGAZINE, vol. 32, no. 5, 1 May 1994, pages 68-80, XP000451097 CHANG Y -H: "AN OPEN-SYSTEMS APPROACH TO VIDEO ON DEMAND"	1	
A	* the whole document *	2,3	
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	* the whole document *		TECHNICAL FIELDS SEARCHED (Int.Cl.6)
			H04N
A	EP 0 363 081 A (GEN INSTRUMENT CORP) 11 April 1990 * abstract *	1	
The present search report has been drawn up for all claims			
Place of search BERLIN		Date of completion of the search 19 February 1997	Examiner Greve, M
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